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Analysis of nonlinear behavior of loudspeakers using the instantaneous frequency
Abstracts of papers

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Session 2aAA

Architectural Acoustics and Engineering Acoustics: Multi-Channel Sound Reinforcement Systems

Robert C. Coffeen, Chair
School of Architecture and Urban Design, University of Kansas, Marvin Hall, Lawrence, Kansas 66045

Chair's Introduction—8:00

Invited Papers

8:05

2aAA1. Systems for stereophonic sound reinforcement. Jim Brown (Audio Systems Group, Inc., 4875 N. Ravenswood, Chicago, IL 60640)

Although stereophonic loudspeaker systems for large spaces were developed in well-documented work at Bell Labs nearly 70 years ago, most contemporary designers appear to be ignorant of the critically important basic principles as applied to modern reinforcement systems. In fact, many designers don't believe stereo can even work in a room larger than a living room. This paper draws on both the literature and the author's experience over nearly 20 years with both permanent and portable systems using two and three front-referenced channels. Design criteria and examples of both good and problematic design practices are presented.

8:25

2aAA2. Providing audible information relating to the differences in monophonic and multichannel sound reinforcement systems using computer auralization. Robert C. Coffeen (School of Architecture & Urban Design, Marvin Hall, The Univ. of Kansas, Lawrence, KS 66045)

When selecting the sound distribution system for a particular venue, there is often a decision to be made as to the audible differences between a monophonic and potentially less expensive loudspeaker system and a multi-channel and potentially more expensive system. Computer auralization is a tool that can be used for auditioning the approximate differences between single channel and multi-channel loudspeaker systems.

8:45

2aAA3. Using 3-D modeling to design multichannel audio systems. Bruce C. Olson (Olson Sound Design, 8717 Humboldt Ave. N., Brooklyn Park, MN 55444, BCO@OlsonSound.com)

Implementation of multichannel sound systems requires a balancing of a number of different parameters and criteria to produce a cohesive sound field that is appropriate to the venue. Use of a 3-D modeling application allows for an iterative and interactive approach to optimizing the system. This design process will be explored and explained using some recently completed projects. Coordination with specific acoustical features of the room is inherently part of this process.

9:05

2aAA4. Microphone and production techniques for stereophonic sound reinforcement. Jim Brown (Audio Systems Group, Inc., 4875 N. Ravenswood, Chicago, IL 60640)

Producing a stereo image for many listeners in a large room is a far more demanding task than doing so for a few listeners in a living room, and very different acoustic systems of both microphones and loudspeakers are required for large room listening as compared to small room listening. As long ago as the 1930s, William B. Snow and his team at Bell Labs clearly understood these diffferences. Their thoughts on the subject are very different from those of Blumlein, and are well documented. Nearly all thinking and writing about microphone technique since Snow has focused on recording and playback for a single listener centered between two or three loudspeakers in a small room. Stereophonic reinforcement imposes a very different set of requirements, in that both listeners and performers are distributed over a wide area, and the system is a closed loop (that is, the potential for acoustic feedback must be considered).

Contributed Papers

9:25

2aAA5. Derivation of moving-coil loudspeaker parameters using acoustical testing techniques: Theoretical developments. Timothy W. Leishman and Brian E. Anderson (Dept. of Phys. and Astron., Brigham Young Univ., N335 Eyring Sci. Ctr., Provo, UT 84602, tim_leishman@byu.edu)

Moving-coil loudspeaker driver parameters are generally derived through the measurement of electrical impedances. Nevertheless, because these drivers are electro-mechano-acoustical transducers, their parameters may also be determined from measurements taken in other physical domains. This paper presents theoretical concepts that show how they may be determined acoustically using plane wave tube techniques. A driver is mounted in a baffle to form a composite partition between a source tube and a receiving tube. The frequency-dependent transmission loss of the partition is determined using upstream and downstream sound field decompositions that compensate for possible nonanechoic receiving tube conditions. A transmission loss curve based on an equivalent circuit model of the system is then fit to the measured curve to extract specific driver parameters. Different electrical conditions are imposed at the driver terminals to modify the transmission loss in ways that allow the determination of additional parameters.

9:40

2aAA6. Derivation of moving-coil loudspeaker parameters using acoustical testing techniques: Experiment results. Brian E. Anderson and Timothy W. Leishman (Dept. of Phys. & Astron., Brigham Young Univ., N281 ESC Brigham Young Univ., Provo, UT 84602, LoudspeakerDesign@hotmail.com)

A unique acoustical method of measuring small-signal moving-coil loudspeaker parameters has recently been developed. This technique involves the use of a plane wave tube to measure acoustical properties (e.g., reflection and transmission coefficients) of a driver under test (DUT). From this data, small-signal parameters are derived using curve-fitting techniques. Electrical conditions are easily controlled and automated to allow for the derivation of additional parameters. Current parameter measurement techniques require measurement of the electrical impedance of the DUT. This paper will discuss the acoustical measurement apparatus, experimental measurement techniques, and compare its measured parameters to those derived using electrical techniques.

9:55

2aAA7. Analysis of nonlinear behavior of loudspeakers using the instantaneous frequency. Hai Huang (College of Biomed. Eng. and Instrumentation Sci., Zhejiang Univ., Hangzhou 310027, PROC, hhai@zjuem.zju.edu.cn) and Finn Jacobsen (Tech. Univ. of Denmark, DK-2800 Lyngby, Denmark)

It is well know that the weakest link in a sound reproduction chain is the loudspeaker. The most significant effect on the sound quality is nonlinear distortion of loudspeakers. Many methods are applied to analyze the nonlinear distortion of loudspeakers. Almost all of the methods are based on the Fourier transform. In this work, a new method using the instantaneous frequency is introduced for describing and characterizing loudspeaker nonlinearities. First, numerical integration is applied to simulate the nonlinearities of loudspeakers caused by two nonlinear parameters, force factor and stiffness, separately. Then, a practical loudspeaker is used in an experiment and its nonlinear characteristics are analyzed with the instantaneous frequency. The results provide a clear physical interpretation of the nonlinearities of loudspeakers and will be useful for understanding the nonlinear behavior of loudspeakers. It will also be helpful for compensating for the nonlinearities and for improving the quality of sound reproduction. [Work supported by Sino-Danish International Co-operative Project, No. AM13: 66 and DANIDA (Denmark).]

10:10–10:25 Break

10:25

2aAA8. Real-time control of sound diffusion parameters. Colby Leider (School of Music, Univ. of Miami, 1314 Miller Dr., Coral Gables, FL 33124)

Much electronic and computer music relies extensively on real-time diffusion of electronic sound in surround loudspeaker configurations. The most common method of sound diffusion in practice today is to use standard sound-reinforcement mixers whereby each fader controls the playback volume of a corresponding loudspeaker in a concert hall. Using the mixer this way, however, presents problems when one attempts to create sound trajectories, because complex, precise, and repeatable control of the individual faders is required. To address these interface issues, two different handheld controllers were created for real-time sound diffusion. The first controller is equipped accelerometers, force-sensing resistors, and joysticks and maps using input into diffusion parameters. The second controller mimics the operation of a standard mixer but allows more rapid movement of fader positions by replacing each fader with a bend sensor. To test the validity of each controller, twenty undergraduate music engineering students were asked to repeatedly perform various diffusion tasks (such as moving a monaural sound in a circle around the audience) using a standard mixer and each of the controllers. The accuracy and speed of their performances were tracked. The study concludes with lessons learned from the statistics gathered. [Research supported by the University of Miami.]

10:40

2aAA9. The virtual microphone technique in active sound field control systems. Iraklis E. Lampropoulos and Yasushi Shimizu (Prog. in Architectural Acoust., School of Architecture, Rensselaer Polytechnic Inst., 110 8th St., Troy, NY 12180-3590, lampri@rpi.edu)

Active Sound Field Control (AFC) has been proven very useful in reverberation enhancement applications in large rooms. However, feedback control is required in order to eliminate peaks in the frequency response of the system. The present research closely follows the studies of Shimizu in AFC, in which smoothing of the rooms transfer function is achieved by averaging the impulse responses of multiple microphones. ''The virtual or rotating microphone technique'' reduces the number of microphones in the aforementioned AFC technology, while still achieving the same acoustical effects in the room. After the impulse responses at previously specified pairs of microphone positions are measured, the ratio of transfer functions for every pair is calculated, thus yielding a constant K. Next, microphones are removed and their impulse responses are reproduced by processing the incoming signal of each pair through a convolver, where the computed K constants have been previously stored. Band limiting, windowing and time variance effects are critical factors, in order to reduce incoherence effects and yield reliable approximations of inverse filters and consequently calculations of K. The project is implemented in a church lacking low frequency reverberation for music and makes use of 2 physical and 2 virtual microphones.

10:55

2aAA10. Analysis of multiple listener equalization performance due to displacement effects. Sunil Bharitkar, Philip Hilmes, and Chris Kyriakakis (Immersive Audio Lab., EEB 428, Integrated Media Systems Ctr., Univ. of Southern California, Los Angeles, CA 90089, bharitka@usc.edu)

Traditionally, room response equalization is performed to enhance sound quality, by reducing the effects of reverberation, at a given listener. However, room responses vary with source and listener positions. Hence, in a multiple listener environment, equalization may be performed through averaging the room responses measured at multiple listener locations. However, the performance of averaging based equalization, at the listeners, may be affected when listener positions change or due to microphone-listener position mismatch (i.e., displacement effects). In this paper, we present a statistical approach to map variations in listener positions to equalization performance of spatial average methods. The analysis is done at frequencies above the Schroeder frequency where the direct and the reverberant sound fields are uncorrelated, and the results are presented in a 3-D plot to clearly show the changes in equalization performance (in dB's) versus mismatch and frequencies for various listener configurations relative to a fixed loudspeaker source. The results indicate that, for the analyzed listener configurations, the zone of equalization depends on distance of microphones/listeners from the source and the frequencies in the sound. [Work supported by the U.S. Dept. of Army.]

11:10

2aAA11. Open-loop dereverberation of multichannel room impulse responses. Bowon Lee, Mark A. Hasegawa-Johnson (Dept. of Elec. and Computer Eng., Univ. of Illinois at Urbana–Champaign, 1406 W. Green St., Urbana, IL 61801, bowonlee@uiuc.edu), and Camille Goudeseune (Integrated Systems Lab., Beckman Inst., Urbana, IL 61801)

We are developing the audio display for a CAVE-type virtual reality theater, a 3-m cube with displays covering all six rigid faces. The user's headgear continuously reports ear positions so headphones would be possible, but we nevertheless prefer loudspeakers because this enhances the sense of total immersion. Because sounds produced at the loudspeakers are distorted by the room impulse responses, we therefore face the problem of controlling the sound at the listener's two ears. Our proposed solution consists of open-loop acoustic point control, i.e., dereverberation. The room impulse responses from each loudspeaker to each ear of the listener are inverted using multichannel inversion methods, to create exactly the desired sound field at the listener's ears. Because the actual room impulse responses cannot be measured in real time (as the listener walks

around), instead the impulse responses simulated by the image-source method is used. A new evaluation criterion is proposed to quantitatively evaluate both the simulation and the open-loop dereverberation. The actual impulse responses used for this evaluation are measured with a starter pistol, since this best approximates the point source assumed by the image-source method.

11:25

2aAA12. Multichannel sound reinforcement systems at work in a learning environment. John Malek and Colin Campbell (10489 E. Grand River Ste. I, Brighton, MI 48116, jmalek@annarbaud.com)

Many people have experienced the entertaining benefits of a surround sound system, either in their own home or in a movie theater, but another application exists for multichannel sound that has for the most part gone unused. This is the application of multichannel sound systems to the learning environment. By incorporating a 7.1 surround processor and a touch panel interface programmable control system, the main lecture hall at the University of Michigan Taubman College of Architecture and Urban Planning has been converted from an ordinary lecture hall to a working audiovisual laboratory. The multichannel sound system is used in a wide variety of experiments, including exposure to sounds to test listeners' aural perception of the tonal characteristics of varying pitch, reverberation, speech transmission index, and sound-pressure level. The touch panel's custom interface allows a variety of user groups to control different parts of the AV system and provides preset capability that allows for numerous system configurations.

11:40

2aAA13. Virtual AM stereo and surround sound to setup AM/FM radio theatre. Selvakumaran Vadivelmurugan, K. Veeraraghavan (Dept. of I.T., Sri Venkateswara College of Eng., Anna Univ., Pennalur, Sriperumbudur, Tamil Nadu, India 602105, vselvakumaran@vselvakumaran.com), and Sharavan V. Govindan (Univ. of Connecticut, Storrs, CT 06269)

Introduction of virtual surround sound and stereo to AM radio has been proposed in this study. This technology can be further applied to aid the construction of an AM radio theatre. Adding to the advantages of AM, the lower bandwidth, higher range, and simpler circuitry, AM can now offer excellent sound effect with the post-transmission process. The motivation for the introduction of virtual surround sound is the poor quality of AM sound. In this study, the response by human ear has been thoroughly investigated and the methodology to create virtual surround sound has been developed. The elements essential to setup audio theatre such as the components of audio chain, multiple unit audio speaker, inner section of the ear, psychological effect of different ranges of frequencies, and radio theatre design have been extensively studied on the basis of Helmholtz audition theory. The vital changes include the different frequency division multiplexing of message at the transmitting end, three phases of the process, resulting in the vertical and horizontal digital connection, espresso program, and the 3×12 speaker design theatre.

TUESDAY MORNING, 29 APRIL 2003 ROOMS 103/104, 8:00 TO 10:00 A.M.

Session 2aAO

Acoustical Oceanography, Underwater Acoustics and Signal Processing in Acoustics: Geoacoustic Inversion II

N. Ross Chapman, Chair

School of Earth and Ocean Sciences, University of Victoria, P.O. Box 3055, Victoria, British Columbia V8W 3P6, Canada

Invited Papers

8:00

2aAO1. Modal inverse methods: An overview. Kyle M. Becker (Appl. Res. Lab., Penn State Univ., P.O. Box 30, State College, PA 16804-0030) and George V. Frisk (Woods Hole Oceanogr. Inst., Woods Hole, MA 02543)

Geoacoustic inversion using modal inverse methods is based on a very simple experimental geometry. Data are acquired from synthetic aperture measurements of a point source acoustic field using a single hydrophone translating through the water column at constant depth. The modal composition of the propagating field is extracted by exploiting the Hankel transform relationship between the acoustic field, measured as a function of range, and the horizontal wavenumber spectrum for horizontally stratified media. Modal content, estimated as discrete values of the horizontal wavenumber, are used as input data for a general linear inverse problem, where small perturbations to a background sound speed profile are related to changes in individual horizontal wavenumbers. An overview of the method is presented with an emphasis placed on the application to range-dependent shallow water environments. Inversion results are presented from a recent workshop on geoacoustic inversion for range-dependent environments, as well as from recent at-sea measurements. [Work supported by ONR.]

8:20

2aAO2. Estimating the low-frequency (0.1–1 kHz) sound speed in marine sediments using the harmonics from the propeller of a light aircraft. Michael J. Buckingham and Eric M. Giddens (Marine Physical Lab., Scripps Inst. of Oceanogr., 8820 Shellback Way, La Jolla, CA 92093-0238)

During ONR's Sediment Acoustics Experiment 1999 (SAX99) in the northeastern Gulf of Mexico, several research groups made high-precision, *in situ* measurements of dispersion in the medium-sand sediment at frequencies greater than 20 kHz. Comparable precision at lower frequencies is difficult to achieve with *in situ* time-of-flight techniques because of wavelength issues which, *inter alia*, dictate an inconveniently large and costly acoustic source. Yet low-frequency (1 kHz) sound speed measurements are sorely needed to distinguish between competing theoretical predictions. An alternative to the traditional travel-time approach employs a single hydrophone buried in the sediment and, instead of an *in situ* sound source, the low-frequency harmonics from the propeller of

a light aircraft. Essentially, the airborne-source technique relies on the difference between the Doppler-shifted frequencies on aircraft approach and departure, as detected on the buried hydrophone, to yield a direct measure of the local sound speed at the detector. Experiments recently conducted about 1.5 km north of Scripps pier, using a single-engine Tobago TB10 aircraft with a two-blade propeller, will be described and the resultant estimates of the low-frequency sound speed in the fine-sand sediment at the site will be presented. [Work supported by ONR.]

8:40

2aAO3. Inversion of sediment property using ambient noise. Dajun Tang (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105)

Recently, there have been a couple of groups started using ambient noise and a vertical array to invert for bottom geo-acoustic properties. The basic idea is to select beam outputs from the array of the noise field which is generated near the ocean surface. The ratio of the two beams with angles symmetrical to the horizontal gives the reflection loss of the bottom. The reflection loss can be obtained in large and small grazing angles and over a frequency band limited only by the array length and spacing. In this paper, both modeling and data are presented to demonstrate the applicability and limitations of the approach. Emphasis will be on the resolution and uncertainty of the method when the array is of finite length and has a tilt angle relative to the vertical. The noise field will be generated using numerical methods based on Kraken. Wind as well as shipping noise is considered.

Contributed Papers

9:00

2aAO4. Seabottom geoacoustic inversion from reverberation vertical coherence in shallow water. Ji-Xun Zhou and Xue-Zhen Zhang (School of Mech. Eng., Georgia Inst. of Technol., Atlanta, GA 30332-0405 and Inst. of Acoust., Chinese Acad. of Sci., Beijing 100080, PROC, jixun.zhou@me.gatech.edu)

Shallow-water reverberation from one shot offers a continuous spatial sampling of the surrounding sound field up to several tens of kilometers from the source. It involves a bottom-controlled two-way sound propagation and bottom scattering process, and brings back rich information on seabottom geoacoustic parameters. Thus, geoacoustic inversion from reverberation in shallow water is very attractive for saving time and cost compared with inversion from propagation measurements. In this paper, the reverberation vertical coherence in shallow water, derived by ray-mode analogies [Zhou, Acta Oceanol. Sinica **1**, 212–218 (1979)], is converted back to a more familiar summation of normal-modes. Measured reverberation cross-correlation coefficients as a function of time, frequency and hydrophone separation at different areas with a flat seabed are in good agreement with theoretical predictions. From the best match between the measurements and predictions of the reverberation vertical coherence, sound velocities and attenuations in sediments from China seas, including the ASIAEX01 site, have been inverted at low and mid-frequencies. [Work supported by ONR and CAS.]

9:15

2aAO5. Source depth and array tilt effects on seabed inversion of ambient noise. Francine Desharnais, David J. Thomson (DRDC Atlantic, P.O. Box 1012, Dartmouth, NS B2Y 3Z7, Canada), and Chris A. Gillard (Defence Sci. and Technol. Organisation, Salisbury SA 5108, Australia)

Ambient noise coherence between two vertically separated sensors in shallow water relates to the directionality of the noise field and is sensitive to the reflective properties of the surficial sediments. In earlier work, an energy-flux model was developed to calculate the noise coherence over a multilayered seabed; in this model the surface sources were assigned a frequency-independent dipole directivity pattern. The model was subsequently combined with a hybrid nonlinear inversion procedure to effectively search the space of geoacoustic properties that parametrize a multilayered seabed. In order to benchmark the energy-flux model against a wave-theoretical formulation, the issue of the unknown but finite source depth of the noise sources must be addressed. These near-surface sources result in a frequency-dependent beam pattern that is not dipolar above a few hundred hertz [M. J. Buckingham and N. M. Carbone, J. Acoust. Soc. Am. **102**, 2637–2644 (1997)]. In this paper, we extend the energy-flux model to include the effect of finite source depth as well as sensor tilt on the observed directivity of the ambient noise field. These extra degrees of freedom are added to the search space of the inversion procedure and their influence on estimating seabed properties from noise coherence data is investigated.

9:30

2aAO6. Depth partitioning of modal energy of guided acoustic waves in shallow water as an additional input for geoacoustic inversion. Allan D. Pierce, George V. Frisk (Woods Hole Oceanogr. Inst., Woods Hole, MA 02543), and Luiz L. Souza (MIT, Cambridge, MA 02139)

Modal mapping experiments (MOMAX) previously described by Souza, Frisk, and Becker [J. Acoust. Soc. Am. **112**, 2281 (2001)] employ a CW-source at a fixed water depth and a receiver also at a fixed depth, with the range between the two varying systematically. The received amplitude versus range over a limited range interval yields a *k*-space spectrum, where the individual peaks correspond to the eigen-wavenumbers of the individual guided modes. This extraction in conjunction with a knowledge of the sound speed profile within the water column yields discrete values of the impedance at the water-bottom interface for the corresponding frequency and horizontal wavenumber, and these values in turn can be used in an approximate geoacoustic inversion for the bottom properties. The areas under these peaks are shown to be especially robust and can be given a theoretical interpretation with reference to the modal sum solution for a point source in shallow water with weak range dependence. This yields the ratio of the modal energies in the water column to those in the bottom. Both types of information are combined into a theory that yields an improved geoacoustic inversion.

9:45

2aAO7. Autoregressive wave number estimation technique for range-dependent shallow-water waveguides with abrupt environmental variations. Luiz L. Souza (MIT/WHOI Joint Prog. in Oceanogr./Appl. Ocean Sci. and Eng., 77 Massachusetts Ave., Rm. 5-435, Cambridge, MA 02139) and George V. Frisk (Woods Hole Oceanogr. Inst., Woods Hole, MA 02543)

We extend the technique described by Becker and Frisk [J. Acoust. Soc. Am. **111**, 2388 (2002)], which uses an autoregressive spectral (AR) estimator to infer characteristic modal wave numbers in shallow-water, range-dependent waveguides. Their method offers high resolution over relatively small range apertures, an improvement over the short-range Fourier transform technique originally employed. When the environment varies continuously, e.g., due to a sloping bottom, the assumption of constant modal eigenvalues over a range aperture, intrinsic to the AR estimator, becomes invalid, resulting in increased bias. Abrupt changes in the modal content of the field can be detected, but not measured. The present work extends this method by allowing the modal structure to change with range on a sample by sample basis, and yields AR coefficients that are functions of range, analogous to the time-varying AR (TVAR) estimators used for nonstationary time series analysis. By processing the data twice, with increasing and decreasing ranges (forward and backwards), and combining the results, both slow and abrupt changes in the eigenvalues can be measured.

TUESDAY MORNING, 29 APRIL 2003 ROOM 208, 11:00 A.M. TO 12:05 P.M.

Session 2aBB

Biomedical Ultrasound/Bioresponse to Vibration and ASA Committee on Archives and History: History of Bioresponse to Vibration in the Acoustical Society of America

Stanley J. Bolanowski, Chair
Institute for Sensory Research, Syracuse University, Syracuse, New York 13244

Chair's Introduction—11:00

Invited Paper

11:05

2aBB1. History of bioresponse to vibration in the Acoustical Society of America. Janet M. Weisenberger (Speech & Hearing Sci., Ohio State Univ., Columbus, OH 43210, jan+@osu.edu)

Human response to vibratory stimulation of the skin surface has long been considered an aspect of the sense of touch; however, the debate over whether vibration was one aspect of pressure sensation, as espoused by von Frey in the late 1800s, or a separate sense, as argued by Katz (1925), focused attention on this mode of stimulation. Experimental investigations from the 1920s to the 1960s by Knudsen, Geldard, Sherrick, Verrillo, Mountcastle, and others provided basic data on vibrotactile perception and the neural transduction of vibratory stimulation. Within the Acoustical Society of America, work on bioresponse to vibration has included not only basic investigations of vibrotactile perception and physiology, but also studies of the loss of sensitivity resulting from intense hand-arm vibration induced by occupational use of chainsaws and jackhammers, studies of human response to whole-body vibration, and evaluations of the utility of vibrotactile devices for communication of speech to hearing-impaired persons. Contributions in each of these areas, as well as future research directions, are discussed.

2a TUE. AM

TUESDAY MORNING, 29 APRIL 2003 ROOMS 209/210, 7:25 A.M. TO 12:30 P.M.

Session 2aPA

Physical Acoustics: Sono(con)-fusion I: Evaluating the Chances and Claims of Bubble Fusion

D. Felipe Gaitan, Cochair
Impulse Devices, Inc., 12731-A Loma Rica Drive, Grass Valley, California 95945

R. Glynn Holt, Cochair
Aerospace and Mechanical Engineering, Boston University, 110 Cummington Street, Boston, Massachusetts 02215

Thomas J. Matula, Cochair
Applied Physics Laboratory, University of Washington, 1013 N.E. 40th Street, Seattle, Washington 98105

Chair's Introduction—7:25

Invited Papers

7:30

2aPA1. Update and clarifications on analytic studies for nuclear emissions during acoustic cavitation. Robert I. Nigmatulin (Russian Acad. of Sci., Russia), R. T. Lahey (Rensselaer Polytechnic Inst., Troy, NY), R. P. Taleyarkhan, and C. D. West (Oak Ridge Natl. Lab., Oak Ridge, TN 37831)

A one-dimensional hydrodynamic shock (HYDRO) code was developed to numerically evaluate the conservation equations of each phase during bubble growth and collapse. This code includes the Mie–Gruniesen equations of state and Born–Mayer potential functions, which are known to be valid for highly compressed fluids. In particular, for acetone these equations of state are based on

the shock wave adiabat data, and they implicitly specify the effect of the induced radiation field and the dissociation and ionization processes that take place during plasma formation within imploding bubbles. Moreover, relevant energy losses and the effect of both molecular and electron conductivity were taken into account, and the resultant HYDRO code allowed for the evaluation of shock wave interaction using the well-established Godunov numerical technique. Bubble dynamics were studied in deuterated acetone for conditions typical of those in our experiments. It was found that highly compressed conditions suitable for thermonuclear fusion were predicted, and the results were sensitive to the values of the phase change (that is, accommodation) coefficient, a, and the liquid pool temperature To.

8:15

2aPA2. Why seek fusion from cavitation: Molecular dynamic simulations and a detector capable of time correlated single neutron counting. Carlos Camara, Robert Cousins, Brian Naranjo, Seth Putterman (Phys. Dept., UCLA, Los Angeles, CA 90095), Barry Merriman (UCLA, Los Angeles, CA 90095), and Steven Ruuth (Simon Fraser Univ., Burnaby, BC V5A 1S6, Canada)

The blackbody spectra, and similar sonoluminescence intensities of He and Xe bubbles suggest that the interior of a sonoluminescing bubble is highly stressed and dense. Molecular dynamic simulations indicate interior temperatures which are enhanced by thermal conduction and can approach 1 MK. Furthermore the gas passes through states where the mean free path is larger than the distance over which temperature varies and so calls into question the value of theories based on hydrodynamics. To search for rare fusion events a neutron detector with 25% total discriminated quantum efficiency has been built. It can time stamp neutron arrival and sonoluminescence to better than 1 ns and record tracks on the fly. [Work supported by DARPA.]

9:00–9:15 Break

9:15

2aPA3. Acoustically driven spherical implosions and the possibility of thermonuclear reactions. D. Felipe Gaitan and Ross Tessien (Impulse Devices, 12731-A Loma Rica Dr., Grass Valley, CA 95945)

Acoustically driven, gas-filled cavities in liquids have been known to collapse violently, generating short flashes of light of ~100-ps duration. More recently, the possibility of generating fusion reactions using acoustics (acoustic inertial confinement fusion) has been considered. Results of computer simulations using the HYADES hydrocode (Cascade Applied Sciences, Inc) plus the SESAME equations of state for free collapsing and acoustically driven cavities in molten metals will be presented as well as experimental data at high ambient pressures in different liquids. Back-of-the-envelope calculations in terms of the acoustical and thermodynamic parameters necessary to achieve thermonuclear reactions will be presented in an effort to evaluate the feasibility of acoustic ICF as an energy source.

10:00

2aPA4. Chemical control of single bubble cavitation. Yuri T. Didenko and Kenneth S. Suslick (Univ. of Illinois at Urbana–Champaign, 600 S. Mathews Ave., Urbana, IL 61801, ksuslick@uiuc.edu)

Sonochemistry would be ideally studied with a single bubble with known size pulsating in known acoustic pressure field. Single bubble cavitation provides the means to make such studies. The promise that single bubble cavitation brought to the quantitative measurements of chemical activity of cavitation, however, has not been previously fulfilled due to the very small amount of reacting gas within a single bubble (typically $<$10–13 moles). We have now quantitated the chemical reactions inside a single cavitating bubble and established an inventory of energy dissipation during bubble collapse. The yields of nitrite ions, hydroxyl radicals, and photons from a single cavitation bubble have now been measured, and the first true measures of energy efficiencies during acoustic cavitation have been determined. The energy efficiency of the formation of hydroxyl radicals from single bubble is comparable to that in multibubble cavitation. The energy efficiency of light emission, however, is much higher for SBSL. The observed rate of nitrite formation is in good agreement with the calculated diffusion rate of nitrogen into the bubble. The temperatures reached during single bubble cavitation in liquids with significant vapor pressures will be substantially limited by the endothermic chemical reactions of the polyatomics inside the collapsing bubble.

10:45

2aPA5. Litho-Fusion and HIFUsion: Alternative ways to generate hot bubbles? Thomas Matula, Paul Hilmo, Michael Bailey, and Lawrence Crum (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105)

The recent sonofusion experiments at Oak Ridge National Laboratory utilized standing acoustic wave fields that produced negative pressures of about 15 atm. The generated bubbles were thought to grow from around 100 nm to a maximum size of about 1 mm, before collapsing violently. There are other means for generating bubbles that grow to millimeter sizes. For example, in lithotripsy, a focused shock wave (Pa$\sim -$100 atm) creates bubbles that grow to millimeter sizes. High-intensity focused ultrasound (HIFU) pulses can also generate large bubbles. Can these systems be used to investigate extreme bubble collapse violence? We will describe our efforts in generating and observing bubbles from these systems. If time permits, we will also point out that bubbles in a standing wave undergo translational motion, and this motion may lead to instabilities.

11:30–12:30

Panel Discussion

Session 2aPP

Psychological and Physiological Acoustics: Developmental Psychoacoustics

Lynne A. Werner, Cochair
Department of Speech and Hearing, University of Washington, 1417 NE 42nd Street, Seattle, Washington 98105-6246

Prudence Allen, Cochair
Department of Communication Disorders, University of Western Ontario, 1500 Elborn College, London, Ontario N6G 1H1, Canada

Chair's Introduction—8:30

Invited Papers

8:35

2aPP1. Development of infants' pitch perception. Marsha G. Clarkson (Dept. of Psych., Georgia State Univ., Atlanta, GA 30303)

Several studies demonstrate that by 7 months of age infants can hear the pitch of the missing fundamental for tonal complexes and that the same acoustic cues used by adults permit infants to perceive pitch. Infants and adults were tested in an operant conditioning procedure and learned to categorize tonal complexes with differing spectra according to the pitch of the missing fundamental. Responses to spectral manipulations that reduce the salience of pitch suggested that the strength of the pitch percept may be weaker and the use of temporal cues may be poorer for infants than for adults. Other work utilizing an iterated rippled noise stimulus (IRN) attempted to quantify these developmental differences. Applying progressively greater attenuation to the delayed noise in an IRN progressively decreases the strength of the pitch percept evoked by the stimulus. The maximum attenuation for which listeners detect changes in pitch estimates the strength of their pitch percept. Such thresholds revealed that the strength of the pitch percept evoked by IRN was significantly weaker for infants (6.7 dB) than for adults (19.1 dB). In combination, these findings suggest that temporal mechanisms for processing pitch may be particularly immature in infants.

9:00

2aPP2. Speech intelligibility in complex acoustic environments in young children. Ruth Litovsky (Waisman Ctr., Univ. of Wisconsin–Madison, Madison, WI 53705)

While the auditory system undergoes tremendous maturation during the first few years of life, it has become clear that in complex scenarios when multiple sounds occur and when echoes are present, children's performance is significantly worse than their adult counterparts. The ability of children (3–7 years of age) to understand speech in a simulated multi-talker environment and to benefit from spatial separation of the target and competing sounds was investigated. In these studies, competing sources vary in number, location, and content (speech, modulated or unmodulated speech-shaped noise and time-reversed speech). The acoustic spaces were also varied in size and amount of reverberation. Finally, children with chronic otitis media who received binaural training were tested pre- and post-training on a subset of conditions. Results indicated the following. (1) Children experienced significantly more masking than adults, even in the simplest conditions tested. (2) When the target and competing sounds were spatially separated speech intelligibility improved, but the amount varied with age, type of competing sound, and number of competitors. (3) In a large reverberant classroom there was no benefit of spatial separation. (4) Binaural training improved speech intelligibility performance in children with otitis media. Future work includes similar studies in children with unilateral and bilateral cochlear implants. [Work supported by NIDCD, DRF, and NOHR.]

9:25

2aPP3. Developmental MLD effects. Emily Buss, Joseph W. HallIII, and John H. Grose (Univ. of North Carolina at Chapel Hill, Chapel Hill, NC 27514)

Whereas the masking-level difference (MLD) in children is similar to that found in adults for wideband masking noise, children often have smaller MLDs for narrowband noise maskers. One possibility is that the small MLD of children in narrowband noise is related to the pronounced amplitude fluctuation of the masker. In the present study, adults and children (aged 5.1 to 10.7 years) were tested in an MLD paradigm in which the detection of brief signals was contrasted for signal placement in masker envelope maxima versus masker envelope minima. Maskers were 50-Hz wide noise bands centered on 500 Hz, and the signals were So or S⟨pi⟩ 30-ms, 500-Hz tones. In agreement with previous studies, it was found that MLDs were greater for masker envelop minima placement than for masker envelope maxima placement. This effect was due to variation across the S⟨pi⟩ conditions. The results indicated that young children are relatively poor in taking advantage of the relatively good signal-to-noise ratios in masker minima when detecting S⟨pi⟩ signals in No maskers. This finding is consistent with the possibility that binaural temporal resolution continues to develop through the first decade of life.

9:50

2aPP4. Learning problems, delayed perceptual development, and puberty. Beverly A. Wright, Steven G. Zecker, and Miriam D. Reid (Dept. of Commun. Sci. and Disord. and Inst. for Neurosci., 2240 Campus Dr., Northwestern Univ., Evanston, IL 60208-3550)

Language-based learning problems affect approximately one person in twelve with no other obvious signs of disorder. Many of these individuals have accompanying deficits in nonlinguistic perception. To determine whether age influences the magnitude of these deficits, thresholds on a set of auditory masking tasks were measured in individuals with learning problems and controls ranging in age from 6 years to adult. Performance improved with increasing age in both groups. However, the thresholds of the individuals with learning problems were most similar to those of controls approximately 2–4 years younger on every task, suggesting that the perceptual development of the affected individuals was delayed by a constant amount. Further, on the subset of conditions on which controls reached adult levels of performance after 10 years of age, the improvement of affected individuals halted at 10 years of age, suggesting that puberty may play a critical role in human perceptual development. Taken together, these data support the idea that some learning problems result from a neuromaturational delay, of unknown breadth, and indicate that neurological changes associated with puberty prevent the complete resolution of delayed perceptual development. [Work supported by NIH/NIDCD.]

10:15–10:30 Break

10:30

2aPP5. Variability and reduced performance of preschool- and early school-aged children on psychoacoustic tasks: What are the relevant factors? Prudence Allen (Natl. Ctr. for Audiol., Univ. of Western Ontario, 2262 Elborn College, London, ON N6G 1H1, Canada)

Young children typically perform more poorly on psychoacoustic tasks than do adults, with large individual differences. When performance is averaged across children within age groups, the data suggest a gradual change in performance with increasing age. However, an examination of individual data suggests that the performance matures more rapidly, although at different times for different children. The mechanisms of development responsible for these changes are likely very complex, involving both sensory and cognitive processes. This paper will discuss some previously suggested mechanisms including attention and cue weighting, as well as possibilities suggested from more recent studies in which learning effects were examined. In one task, a simple frequency discrimination was required, while in another the listener was required to extract regularities in complex sequences of sounds that varied from trial to trial. Results suggested that the ability to select and consistently employ an effective listening strategy was especially important in the performance of the more complex task, while simple stimulus exposure and motivation contributed to the simpler task. These factors are important for understanding the perceptual development and for the subsequent application of psychoacoustic findings to clinical populations. [Work supported by the NSERC and the Canadian Language and Literacy Research Network.]

10:55

2aPP6. Applications of psychophysical models to the study of auditory development. Lynne Werner (Dept. of Speech & Hearing Sci., Univ. of Washington, Seattle, WA 98105-6246)

Psychophysical models of listening, such as the energy detector model, have provided a framework from which to characterize the function of the mature auditory system and to explore how mature listeners make use of auditory information in sound identification. The application of such models to the study of auditory development has similarly provided insight into the characteristics of infant hearing and listening. Infants intensity, frequency, temporal and spatial resolution have been described at least grossly and some contributions of immature listening strategies to infant hearing have been identified. Infants psychoacoustic performance is typically poorer than adults under identical stimulus conditions. However, the infant's performance typically varies with stimulus condition in a way that is qualitatively similar to the adult's performance. In some cases, though, infants perform in a qualitatively different way from adults in psychoacoustic experiments. Further, recent psychoacoustic studies of children suggest that the classic models of listening may be inadequate to describe the children's performance. The characteristics of a model that might be appropriate for the immature listener will be outlined and the implications for models of mature listening will be discussed. [Work supported by NIH grants DC00396 and by DC04661.]

Contributed Papers

11:20

2aPP7. Infants' detection in the presence of masker uncertainty. Lori J. Leibold and Lynne A. Werner (Dept. of Speech and Hearing Sci., Univ. of Washington, Seattle, WA 98105-6246, ljl@u.washington.edu)

Most adults have difficulty detecting a fixed-frequency pure tone when the frequency content of the masker is varied on each presentation. This informational masking is thought to be the result of masker uncertainty. The purpose of the present study was to examine infant thresholds under conditions of masker uncertainty. The subjects were 7- to 9-month-old infants and 18- to 30-year-old adults with no risk factors for hearing loss. Detection thresholds were measured for a 300-ms, 1-kHz tone in the presence of a noise band (300–3000 Hz), a two-component constant-frequency masker, or a two-component random-frequency masker. Maskers repeated at 300-ms intervals throughout the testing session at an overall level of 60 dB SPL. The signal was played synchronously with one interval of the repeating masker. An observer-based testing method was used. Following training to 80%-correct criterion, detection thresholds were determined adaptively. For adults, the random-frequency masker condition produced the greatest amount of masking, ranging from 30–60 dB. For infants, the random-frequency masker also produced the greatest amount of masking, ranging from 60–80 dB. In contrast to the adults, infant thresholds were also elevated in the constant-frequency masker condition, ranging from 35–55 dB across subjects. [Work supported by NIDCD RO1 DC000396 and NRSA DC006122.]

11:35

2aPP8. A cocktail-party listening experiment with children. Frederic Wightman, Michael Callahan, and Doris Kistler (Waisman Ctr., 1500 Highland Ave., Madison, WI 53705)

In an experiment modeled after one reported recently by Brungart and Simpson [J. Acoust. Soc. Am. **112**, 2985–2995 (2002)], 38 children (ages 4–16) and 10 adults responded to a monaural target speech signal in the

presence of one or two distracter speech signals. The target speaker was a male and the distracter speakers were females. When two distracters were present they were in different ears. Performance at several different target ear S/N was measured and psychometric functions were fitted to estimate threshold, or the 50% performance level. The youngest children required approximately 20 dB higher S/N than adults to achieve threshold with a single distracter. This difference disappeared by age 16. The impact of adding the contralateral distracter, which is thought to contribute only informational masking, was roughly constant across age, however. Adult thresholds increased about 11 dB and the thresholds for the youngest children increased about 10 dB. This was surprising given previous experiments that showed much larger informational masking effects in young children. Also inconsistent with previous results is the lack of individual differences. Nearly all listeners showed almost the same contralateral distracter effect. [Work supported by NICHD.]

TUESDAY MORNING, 29 APRIL 2003 ROOM 205, 8:30 TO 11:45 A.M.

Session 2aSC

Speech Communication: Neural Models for Speech Perception

Robert F. Port, Cochair
Department of Linguistics, Indiana University, 330 Memorial Hall, Bloomington, Indiana 47405

Frank H. Guenther, Cochair
Department of Cognitive and Neural Systems, Boston University, 677 Beacon Street, Boston, Massachusetts 02215

Invited Papers

8:30

2aSC1. Introductory remarks on neural modeling in speech perception research. Frank H. Guenther (Dept. of Cognit. and Neural Systems, Boston Univ., Boston, MA 02215, guenther@bu.edu)

This talk will provide an overview of several neural models that have been used to address experimental data on speech perception, including psychophysical and neurophysiological data. Although these models differ greatly in their scope and mathematical detail, all are geared toward providing a better understanding of the neural mechanisms underlying the processing of speech sounds.

8:50

2aSC2. Resonant cortical dynamics of speech perception. Stephen Grossberg (CNS Dept., Boston Univ., 677 Beacon St., Boston, MA 02215)

What is the neural representation of a speech code as it evolves in time? How do listeners integrate temporally distributed phonemic information into coherent representations of syllables and words? How does the brain extract invariant properties of variable-rate speech? This talk describes a neural model that suggests answers to these questions, while quantitatively simulating speech and word recognition data. The conscious speech and word recognition code is suggested to be a resonant wave, and a percept of silence a temporal discontinuity in the rate that resonance evolves. A resonant wave emerges when sequential activation and storage of phonemic items in working memory provides bottom-up input to list chunks that group together sequences of items of variable length. The list chunks compete and winning chunks activate top-down expectations that amplify and focus attention on consistent working memory items, while suppressing inconsistent ones. The ensuing resonance boosts activation levels of selected items and chunks. Because resonance occurs after working memory activation, it can incorporate information presented after intervening silence intervals, so future sounds can influence how we hear past sounds. The model suggests that resonant dynamics enable the brain to learn quickly without suffering catastrophic forgetting, as described within Adaptive Resonance Theory.

9:20

2aSC3. Using neural modeling and functional neuroimaging to study the neural basis of auditory object processing. Barry Horwitz and Fatima T. Husain (Brain Imaging & Modeling Section, NIDCD, Natl. Institutes of Health, Bethesda, MD 20892, horwitz@helix.nih.gov)

The neural basis of auditory object processing in the human cerebral cortex was investigated by combining neural modeling and functional neuroimaging. We developed a large-scale, neurobiologically realistic network model of auditory pattern recognition that relates neuronal dynamics of cortical auditory processing of frequency-modulated (FM) sweeps to functional neuroimaging data obtained using functional magnetic resonance imaging (fMRI). FM sweeps are ubiquitous in animal communication. Areas included in the model extend from primary auditory to prefrontal cortex. The electrical activities of the model neuronal units were constrained to agree with data from the neurophysiological literature regarding FM sweep perception. A fMRI experiment using stimuli and tasks similar to those used in our simulations was performed. The regional integrated synaptic activities of the model were used to determine simulated regional fMRI activities, and generally agreed with the experimentally observed fMRI data. Our results demonstrate that the model is capable of exhibiting the salient features of both electrophysiological neuronal activities and fMRI values that are in agreement with empirically observed data. These findings provide support for our hypotheses concerning how auditory objects are processed by primate neocortex. This type of approach offers the potential for understanding the neural basis of human speech perception.

9:50–10:05 Break

10:05

2aSC4. Modeling the representation of speech sounds in auditory cortical areas. Frank H. Guenther, Jason A. Tourville, and Jason W. Bohland (Dept. of Cognit. and Neural Systems, Boston Univ., Boston, MA 02215, guenther@bu.edu)

Many studies have shown that sounds from near the center of a sound category (such as a phoneme from one's native language) are more difficult to discriminate from each other than sounds from near a category boundary. However, the neural processes underlying this phenomenon are not yet clearly understood. In this talk we describe neural models that have been developed to address experimental data from psychophysical and functional brain imaging experiments investigating sound representations in the cortex. Experiments investigating the effects of categorization and discrimination training with nonspeech sounds indicate that different training tasks have different effects on sound discriminability: discrimination training increases the discriminability of the training sounds, whereas learning a new sound category decreases the discriminability of the training sounds within the category. These results can be accounted for by a neural model in which categorization training causes a decrease in the size of the cortical representation of central sounds in the category, while discrimination training leads to an increase in the cortical representation of training sounds. This model is supported by brain imaging results for speech and nonspeech sounds. Experimental results further suggest preferential utilization of different auditory cortical regions when subjects perform identification versus discrimination tasks.

10:35

2aSC5. The functional anatomy of speech perception: Dorsal and ventral processing pathways. Gregory Hickok (Dept. of Cognit. Sci., Univ. of Calif., Irvine, CA 92697)

Drawing on recent developments in the cortical organization of vision, and on data from a variety of sources, Hickok and Poeppel (2000) have proposed a new model of the functional anatomy of speech perception. The model posits that early cortical stages of speech perception involve auditory fields in the superior temporal gyrus bilaterally (although asymmetrically). This cortical processing system then diverges into two broad processing streams, a ventral stream, involved in mapping sound onto meaning, and a dorsal stream, involved in mapping sound onto articulatory-based representations. The ventral stream projects ventrolaterally toward inferior posterior temporal cortex which serves as an interface between sound and meaning. The dorsal stream projects dorsoposteriorly toward the parietal lobe and ultimately to frontal regions. This network provides a mechanism for the development and maintenance of "parity" between auditory and motor representations of speech. Although the dorsal stream represents a tight connection between speech perception and speech production, it is not a critical component of the speech perception process under ecologically natural listening conditions. Some degree of bi-directionality in both the dorsal and ventral pathways is also proposed. A variety of recent empirical tests of this model have provided further support for the proposal.

11:05

2aSC6. Is there still a TRACE of trace? James McClelland, Daniel Mirman, and Lori Holt (Dept. of Psych. and Ctr. for Neural Basis of Cognition, Carnegie Mellon Univ., Pittsburgh, PA 15213)

According to the TRACE model [McClelland and Elman, Cogn. Psychol. **18**, 1–86 (1986)], speech recognition is an interactive activation process involving the integrated use of top-down (lexical) and bottom-up (acoustic) information. Although it is widely accepted that there are lexical influences on speech perception, there has been a disagreement over their exact nature. Two contested predictions of TRACE are that (a) lexical influences should delay or inhibit recognition of phonemes not consistent with lexical information and (b) a lexical influence on the identification of one phoneme can trigger compensation for co-articulation, affecting the identification of other phonemes. Others [Norris, McQueen, and Cutler, BBS **23**, 299–370 (2000)] have argued that the predicted effects do not occur, taking this to support an alternative to the TRACE model in which lexical influences do not affect perception, but only a post-perceptual identification process. We re-examine the evidence on these points along with the recent finding that lexical information may lead to a lasting adjustment of category boundaries [McQueen, Norris, and Cutler, Psychonomics Abstract **255** (2001)]. Our analysis indicates that the existing evidence is completely consistent with TRACE, and we suggest additional research that will be necessary to resolve unanswered questions.

11:35–11:45

Panel Discussion

Session 2aSP

Signal Processing in Acoustics and Physical Acoustics: Subspace Methods for Acoustical Imaging I

David H. Chambers, Cochair
Lawrence Livermore National Laboratory, L-154, P.O. Box 808, Livermore, California 94551-5508

Sean K. Lehman, Cochair
Lawrence Livermore National Laboratory, L-154, 7000 East Avenue, Livermore, California 94550

Chair's Introduction—9:55

Invited Papers

10:00

2aSP1. Imaging with eigenfunctions of a scattering operator. Robert C. Waag (Depts. of Elec. & Computer Eng. and Radiol., Univ. of Rochester, Rochester, NY 14642)

An inverse scattering method using eigenfunctions of a scattering operator is reviewed. The framework of the method encompasses the use of eigenfunctions for focusing and quantitative image reconstruction in arbitrary media. Scattered acoustic fields are described using an operator. The eigenfunctions correspond to far-field patterns of an effective source distribution. Incident-wave patterns specified by the eigenfunctions focus on the distribution. The eigenfunction focusing properties are employed to reconstruct an unknown scattering medium by using products of numerically calculated fields defined in terms of eigenfunctions. A linearized version of the method is equivalent to the filtered-backprojection method for Born inversion. The full range of frequencies present in an incident pulse waveform can be used in the method. The method is examined by using both calculated and measured data. In the calculations, an exact solution for scattering from nonconcentric cylinders was employed to obtain the scattered field. In the measurements, a novel ring-transducer system was used to obtain the incident and total fields. The results of simulations and experiments show that the method is accurate for objects large compared to the incident pulse center-frequency wavelength and that the point resolution of the method is about one-half the center-frequency wavelength.

10:30

2aSP2. Imaging with the DORT method. Claire Prada, Estelle Kerbrat, Jean-Louis Thomas, and Mathias Fink (Lab. Ondes et Acoustique, ESPCI, 10 rue Vauquelin, 75005 Paris, France)

The decomposition of the time reversal operator (DORT) method applies to active detection and focusing of acoustic waves using arrays of transmitters and receivers. This method allows detection and selective focusing on scatterers through complex media. It consists in the construction of the invariants of the time reversal process, that are also the singular vectors of the array response matrix. The DORT method is particularly interesting for detection in inhomogeneous media when the acoustic properties are poorly known. However, if an estimate of the medium's Green function is available, images of the medium can be formed by backpropagation of the dominant eigenvectors. For the detection of defect in scattering media like titanium, the method is useful to separate the echo of a defect from the microstructure contribution and thus to reduce speckle noise. We will also see that the time reversal operator can be interpreted as a pseudocovariance matrix like those encountered in classical array signal processing. And as shown by Anthony J. Devaney [J. Acoust. Soc. Am. **110**, 2617 (2001)], nonlinear estimators like MUSIC can be applied to achieve high resolution. These different points will be illustrated through several experimental results.

11:00

2aSP3. Vector subspace methods in inverse acoustic wave scattering. Anthony J. Devaney (ECE, Northeastern Univ., 360 Huntington Ave., Boston, MA 02115)

The problems of detecting, locating, and identifying acoustic scatterers embedded in a known inhomogeneous background from the multistatic data matrix collected from an arbitrary unstructured, mixed sensor array are addressed using the distorted wave Born approximation. Based on this formulation two classes of problems are considered: (i) detecting and locating a finite set of point scatterers, (ii) imaging an arbitrary distribution of point or extended scatterers. Problems of the first class are shown to admit solutions based on the singular value decomposition (SVD) of the multistatic data matrix considered as a linear mapping from the finite vector space of transmitter inputs to the finite vector space of receiver outputs. In this case the SVD of the data matrix is shown to lead

directly to generalized time-reversal algorithms that allow super-resolution location estimation. Problems of the second class are also addressed using the SVD but this time it is based on the multistatic data matrix considered as a linear mapping from the Hilbert space of scatterer distributions to the finite vector space of receiver outputs. In this case a generalized form of the filtered backpropagation algorithm is derived and shown to lead to a minimum norm image of the scatterer distribution.

11:30

2aSP4. Remote sensing and communications in random media. George Papanicolaou (Dept. of Math., Stanford Univ., Stanford, CA 94305)

Reliable, high-capacity communications in scattering media can be effectively established with some basic remote sensing techniques involving time reversal. I will formulate these problems and discuss the various mathematical approaches that can be used for analysis. It turns out that stochastic analysis plays an important role and, in some cases, gives very satisfactory results. One such result is the spectacular increase in communications capacity in a richly scattering environment. I will end with a discussion of applications and computational issues that arise in the realistic simulation of communication systems.

TUESDAY MORNING, 29 APRIL 2003 ROOMS 103/104, 10:15 A.M. TO 12:00 NOON

Session 2aUW

Underwater Acoustics, Signal Processing in Acoustics and Engineering Acoustics: Robust Passive Sonar I

Lisa M. Zurk, Cochair
Lincoln Laboratory, Massachusetts Institute of Technology, 244 Wood Street, Lexington, Massachusetts 02173-6426

Brian H. Tracey, Cochair
Lincoln Laboratory, Massachusetts Institute of Technology, 244 Wood Street, Lexington, Massachusetts 02173-6426

Contributed Papers

10:15

2aUW1. Detection of a weak target in the presence of loud moving noise sources. Edmund J. Sullivan (OASIS, Inc., 5 Militia Dr., Lexington, MA 02421), James V. Candy (Lawrence Livermore Natl. Lab., Livermore, CA 94550), and William M. Carey (Boston Univ., Boston, MA 02115)

A previous study [J. Acoust. Soc. Am. **112** (2002)] treated the detection of a weak target in an adverse shallow-water environment with ambient noise and ''known'' interfering ships by applying a model-based adaptive (recursive) technique. The shallow-water environment and noise sources were represented by a normal-mode model directly incorporated into the model-based processor, thereby allowing their effects to be removed from the decision function prior to target detection. This previous work assumed that the interfering ships were motionless. Here, a general state-space framework is developed, that includes a dynamic model of the moving interferers together with the adaptive signal processor in a self-consistent formalism. The decision function is the so-called ''Weighted Sum Square Residual'' (WSSR), which evolves directly from the innovation sequence of the model-based processor. An example is shown for a low level sound source in Gaussian ambient noise with a strongly interfering moving ship. It is shown further that, upon detection, the innovation sequence can be used as ''prefiltered'' data providing input that can be used to track the source after its detection.

10:30

2aUW2. Channel order selection in blind deconvolution based on eigenvector characteristics and using normal modes in conjunction with multipath compression for source identification. James P. LaRue, George E. Ioup, Juliette W. Ioup (Dept. of Phys., Univ. of New Orleans, New Orleans, LA 70148, jlarue@uno.edu), and George B. Smith (Naval Res. Lab., Stennis Space Center, MS 39529)

Two related topics in blind deconvolution are presented. The first concerns channel order selection. Many blind deconvolution techniques assume the filter length to be known. One technique of filter length determination uses the dimension of a nullspace obtained by decomposing correlation matrices. Although the dimension is usually determined by calculations involving the eigenvalues, this alternative method relies on the known characteristics of eigenvectors among the subspaces. Multipath functions in underwater acoustics commonly have high kurtosis. The new method can give correct estimates on filter length for signal-to-noise ratios as low as 10 dB. The matrix methods described may not work as well when the sample length increases. However, the second topic shows how normal mode properties in conjunction with time compression in modeled multipath functions may be used to enhance blind deconvolution in a waveguide when the sample length is large. Given data from a vertical array of hydrophones, one may accurately obtain a source identification. The methods of the tracking and positioning of normal modes are incorporated with an input of a linear frequency-modulated pulse propagated through an ocean environment. [Research supported by ONR.]

10:45

2aUW3. Model-based source localization in the 8–16 kHz band using the channel impulse response function. Paul Hursky, Martin Siderius, Michael B. Porter (Sci. Applications Intl. Corp., 10260 Campus Point Dr., San Diego, CA 92121), and Vincent K. McDonald (Space and Naval Warfare Systems Ctr., San Diego, CA 92152)

SignalEx tests have been conducted in a variety of shallow water coastal environments to relate the performance of acoustic communications systems to the prevailing oceanographic conditions. These tests have typically consisted of using a fixed receiver (one to four elements, spaced for diversity) and a transmitter drifting out to ranges beyond the minimum detectable level. During these tests, waveforms to probe the channel in the 8 to 16 kHz band were transmitted at regular intervals. These probe signals were used to test source localization algorithms at these high frequencies. Model-based source localization at these very high frequencies requires either very accurate modeling or algorithms inherently robust against model mismatch. Although the first couple of multipath arrivals can be stabilized from ping to ping, the later arrivals exhibit rapidly fluctuating amplitudes and times of arrival, due to the motion of the ocean surface, water column variability, and the varying bathymetry as the transmitter drifts in range. Measurements of the time-varying channel response and source localization results using the 8–16 kHz band SignalEx channel probes will be presented from sites at the New England Front and the Coronado Bank off San Diego.

11:00

2aUW4. Horizontal-array beamforming using the waveguide invariant. Daniel Rouseff (Appl. Phys. Lab., College of Ocean and Fishery Sci., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105) and Altan Turgut (Naval Res. Lab., Washington, DC 20375)

It has been shown that with a horizontal array and incoherent processing, the measured acoustic intensity produced by a distant source exhibits striations when plotted versus range and frequency. The striations are a consequence of interference between pairs of propagating acoustic modes, and the waveguide invariant describes their slope. The present work extends this concept to coherent processing, such as beamforming with a horizontal array. Equations are derived that describe the trajectories of the striations observed in LOFARgrams. (LOFARgrams display the time-evolving spectrum of a beamformer for a particular look direction.) The validity of the equations is first tested in numerical simulations and then tested using horizontal-array data from the ONR Asian Seas International Acoustics Experiment (ASIAEx) conducted in the South China Sea. The effect of mismatch between the beamformer look direction and the true bearing of the source is also discussed. [Work supported by ONR.]

11:15

2aUW5. Multipath compression by multiple convolutions. George B. Smith (Code 7185, Ocean Acoust. Div., Naval Res. Lab., Stennis Space Center, MS 39529-5004)

Blind deconvolution algorithms can be useful as pre-processors for signal classification algorithms in shallow water. These algorithms remove the distortion of the signal caused by multipath propagation when no knowledge of the environment is available. A very simple blind algorithm is presented here which utilizes convolutions of multiple data channels for convolutional smoothing of multipath. Computer simulation studies show significant performance improvement when propagation effects are mitigated by the use of this blind technique. [Work supported by ONR and the Naval Research Laboratory.]

11:30

2aUW6. Performance bounds on wave-front curvature ranging. Joseph E. Bondaryk (Titan Corp., 470 Totten Pond Rd., Waltham, MA 02451), Phillip Abbot, Charles Gedney, and Edward Sullivan (Ocean Acoust. Services and Instrumentation Systems, Inc., Lexington, MA 02421)

Acoustic wave-front curvature ranging (WCR) is a localization method that exploits the curvature of the arriving wave-front to determine the range of an acoustic emitter. By assuming that the wave-front is circular, it provides an estimate of the range as the radius of the circle. It is based on the measurement of time of arrival differences for correlated transient signals between pairs of spatially separated receivers. Performance bounds are determined by: (1) source and receiver geometry, (2) signal characteristics, and (3) spatial coherence limits due to oceanic effects. Both theoretical and Monte Carlo studies have been done which define the performance limitations as a function of aperture size, signal structure and spatial coherence. Results show that for a given aperture and fixed range accuracy, the low-frequency limit is bounded by the signal characteristics (SNR and time-bandwidth product), whereas the high-frequency limit is determined by the coherence length allowed by the ocean. This high-frequency limit is presently estimated by extrapolation of lower frequency models and is imprecisely known due to the unavailability of high-frequency, short time spatial coherence data.

11:45

2aUW7. Source bearing estimation using ice-mounted geophones. Stan Dosso (School of Earth and Ocean Sci., Univ. of Victoria, Victoria, BC V8W 3P6, Canada, sdosso@uvic.ca), Michael Vinnins (Defence Res. and Development Canada–Ottawa, Ottawa, ON K1A 0Z4, Canada), and Garry Heard (Defence Res. and Development Canada–Atlantic, Dartmouth, NS B2Y 3Z7, Canada)

This paper presents the results of Arctic field trials carried out to estimate the bearing to acoustic sources in the water column using seismo-acoustic particle motion measured at a tri-axial geophone mounted on the sea ice surface. Source bearings are estimated by applying polarization filters to suppress seismic wave types with transverse particle motion, and computing the incident power rotated into radial look angles from 0–360 deg. The 180-deg ambiguity inherent in this rotational analysis is resolved by requiring prograde particle motion in the radial-vertical plane. Results are presented for smooth and rough annual ice and multiyear ice for source ranges from 200 m to 50 km. The results indicate good bearing estimation to long ranges with little dependence on ice type.

Session 2pAAa

Architectural Acoustics: Student Design Competition

Robert C. Coffeen, Cochair
School of Architecture and Urban Design, University of Kansas, Marvin Hall, Lawrence, Kansas 66045

Lily M. Wang, Cochair
Architectural Engineering, University of Nebraska–Lincoln, 200B Peter Kiewit Institute, 1110 South 67th Street, Omaha, Nebraska 68182-0681

Robin Glosemeyer, Cochair
Jaffe Holden Acoustics, 114A Washington Street, Norwalk, Connecticut 06854

The Technical Committee on Architectural Acoustics of the Acoustical Society of America and the National Council of Acoustical Consultants are sponsoring this Student Design Competition that will be professionally judged at this meeting.

The purpose of this design competition is to encourage students enrolled in architecture, architectural engineering, and other university curriculums that involve building design and/or acoustics to express their knowledge of architectural acoustics and building noise control in the schematic design of a building where acoustical considerations are of primary importance. This competition is open to undergraduate and graduate students from all nations.

The submitted designs will be displayed in this session and they will be judged by a panel of professional architects and acoustical consultants. Up to five entries will be selected for awards, one ''First Honors'' award and four ''Commendation'' awards. An award of $1,000 will be given to the entry judged ''First Honors.'' An award of $500 will be given to each of the entries judged ''Commendation.''

TUESDAY AFTERNOON, 29 APRIL 2003 ROOMS 108/109, 1:30 TO 4:20 P.M.

Session 2pAAb

Architectural Acoustics and Musical Acoustics: Relationships of Synthesis and Processing to ''Acoustical'' Music

K. Anthony Hoover, Chair
Cavanaugh Tocci Associates, Inc., 327F Boston Post Road, Sudbury, Massachusetts 01776

Chair's Introduction—1:30

Invited Papers

1:35

2pAAb1. Virtual acoustics for music practice rooms. Ron Freiheit (Wenger Corp., 555 Park Dr., Owatonna, MN 55050, ron.freiheit@wengercorp.com)

The use of virtual acoustics has provided a new level of practice experience for the musician. By integrating the sound isolation of music practice rooms with the signal processing of an active acoustic system (with time variant-gain before feedback) musicians can now benefit from the experience of practicing in multiple acoustic environments. Musicians select from various acoustics environments from a typical small practice room to that of a large space such as a sports arena. The variability of the acoustic environment allows the musician to hear clearly their intonation and articulation, which may be difficult to discern in a small practice room. To effectively communicate the various acoustics environments, the musicians must be immersed in the sound field of the active acoustics without being able to discern source locations of the speakers. The system must also be able to support the dynamic range of the musicians without presenting artifacts of its own such as system noise or audible distortion. This paper deals with the design constraints needed to meet these requirements as well the antidotal responses from musicians who have used these environments for practice.

1:55

2pAAb2. Using miniature signal processing equipment in real-time brass performance. Thomas J. Plsek (Berklee College of Music, 1140 Boylston St., Boston, MA 02215)

Real time signal processing for brass instrument performance has been in use for more than 20 years now. It has been fraught with many problems not the least of which is the complexity, size, and expense of the equipment as well as the acoustical output of the instrument itself. One device which addresses these issues is the new Yamaha ST5: Personal Studio for brass instruments. By

combining a Yamaha Pickup Mute, which very effectively minimizes the acoustical output of the instrument, with a battery powered unit small enough to be worn on a belt (ca. 5 in.×3 in.×1 in.), this system enables the performer to use such effects as reverb, delay, chorus, equalizer, pitch shifter, etc., that can be used in a wide variety of situations such as private practice, live concert performances, and recordings. By creatively managing the acoustic instrument and the miniature electronic equipment, a reasonably large array of musical resources become available to the performer enabling him/her to enhance existing performance environments, as well as find and develop new ones.

2:15

2pAAb3. The use of delay in multitrack production. Alexander U. Case (Fermata Audio + Acoust., P.O. Box 1161, Portsmouth, NH 03802)

Delay, inevitable whenever sound propagates through space, is too often the bane of the acoustician's practice. An audible echo generally relegates a music performance hall—no matter how beautiful it otherwise might sound—to the lowest status. Multitrack music production on the other hand, with its aggressive use of overdubbing, editing, and signal processing, is not bound by those rules of time and space which determine the sound of a hall. In the recording studio, where music is synthesized for playback over loudspeakers, the delay is employed as a powerful, multipurpose tool. It is not avoided. It is in fact embraced. Echoes are used on purpose, strategically, to enhance the loudspeaker listening experience. Moreover, the humble delay is the basis for many nonecho effects. Flanging, chorus, and pitch shifting are delay-based effects regularly used in audio engineering practice. This paper discusses some of the more common delay-based effects, reviewing their technical structure, the psychoacoustic motivation behind them, and the musical value they create.

2:35

2pAAb4. Music 4C, a multi-voiced synthesis program with instruments defined in C. James W. Beauchamp (School of Music and Dept. of Elec. and Computer Eng., Urbana, IL 61801, j-beauch@uiuc.edu)

Music 4C is a program which runs under Unix (including Linux) and provides a means for the synthesis of arbitrary signals as defined by the C code. The program is actually a loose translation of an earlier program, Music 4BF [H. S. Howe, Jr., *Electronic Music Synthesis* (Norton, 1975)]. A set of instrument definitions are driven by a numerical score which consists of a series of "events." Each event gives an instrument name, start time and duration, and a number of parameters (e.g., pitch) which describe the event. Each instrument definition consists of event parameters, performance variables, initializations, and a synthesis algorithmic code. Thus, the synthetic signal, no matter how complex, is precisely defined. Moreover, the resulting sounds can be overlaid in any arbitrary pattern. The program serves as a mixer of algorithmically produced sounds or recorded sounds taken from sample files or synthesized from spectrum files. A score file can be entered by hand, generated from a program, translated from a MIDI file, or generated from an alpha-numeric score using an auxiliary program, Notepro. Output sample files are in wav, snd, or aiff format. The program is provided in the C source code for download.

2:55–3:05 Break

3:05

2pAAb5. Inside-in, alternative paradigms for sound spatialization. Curtis Bahn and Stephan Moore (Rensselaer Polytechnic Inst., iEAR Studios, Rochester, NY, crb@rpi.edu)

Arrays of widely spaced mono-directional loudspeakers (P.A.-style stereo configurations or "outside-in" surround-sound systems) have long provided the dominant paradigms for electronic sound diffusion. So prevalent are these models that alternatives have largely been ignored and electronic sound, regardless of musical aesthetic, has come to be inseparably associated with single-channel speakers, or headphones. We recognize the value of these familiar paradigms, but believe that electronic sound can and should have many alternative, idiosyncratic voices. Through the design and construction of unique sound diffusion structures, one can reinvent the nature of electronic sound; when allied with new sensor technologies, these structures offer alternative modes of interaction with techniques of sonic computation. This paper describes several recent applications of spherical speakers (multichannel, outward-radiating geodesic speaker arrays) and Sensor-Speaker-Arrays (SenSAs: combinations of various sensor devices with outward-radiating multi-channel speaker arrays). This presentation introduces the development of four generations of spherical speakers—over a hundred individual speakers of various configurations—and their use in many different musical situations including live performance, recording, and sound installation. We describe the design and construction of these systems, and, more generally, the new "voices" they give to electronic sound.

3:25

2pAAb6. The loudspeaker as musical instrument: An examination of the issues surrounding loudspeaker performance of music in typical rooms. David Moulton (Sausalito Audio Works, 61C Galli Dr., Novato, CA, dave@sawonline.com)

The loudspeaker is the most important and one of the most variable elements in the electroacoustic music performance process. Nonetheless, its performance is subject to a "willing suspension of disbelief" by listeners and its behavior and variability are usually not accounted for in assessments of the quality of music reproduction or music instrument synthesis, especially as they occur in small rooms. This paper will examine the aesthetic assumptions underlying loudspeaker usage, the general timbral qualities and sonic characteristics of loudspeakers and some of the issues and problems inherent in loudspeakers interactions with small rooms and listeners.

3:45

2pAAb7. A new loudspeaker design: A case study of an effort to more fully integrate the loudspeaker into the playback room in a musical way. David Moulton (Sausalito Audio Works, 61C Galli Dr., Novato, CA 94949, dave@sawonline.com) and Poul Praestgaard (Bang & Olufsen, DK 7600 Struer, Denmark)

The authors have been members of a design and development team that has created a new loudspeaker that attempts to resolve several of the primary problems presented by the loudspeaker/room/listener interface, as described in one of the author's previous papers. This paper will describe that new loudspeaker, its various new approaches to the interactions between the loudspeaker, the room and the listeners, and a brief review of the research, findings and assumptions underlying its design. The authors hope to have examples of the loudspeaker available for demonstration.

Contributed Paper

4:05

2pAAb8. Physical modeling synthesis of recorder sound. Hiroko Shiraiwa (CCRMA, Dept. of Music, Stanford Univ., Stanford, CA 94305-8180), Kenshi Kishi (Univ. of Electro-Communications, Chofugaoka 1-5-1 Chofu, Tokyo 182-8585, Japan), and Isao Nakamura (Athena Co. Ltd., 1-33-25 Kokuryo-cho, Chofu, Tokyo 182-0022, Japan)

A time-domain simulation of the soprano baroque recorder based on the digital waveguide model (DWM) and an air reed model is introduced. The air reed model is developed upon the negative acoustic displacement model (NADM), which was proposed for the organ flue-pipe simulation [Adachi, Proc. of ISMA 1997, pp. 251–260], based on the semiempirical model by Fletcher [Fletcher and Rossing, *The Physics of Musical Instruments*, 2nd ed. (Springer, Berlin, 2000)]. Two models are proposed to couple DWM and NADM. The jet amplification coefficient is remodeled for the application of NADM for the recorder, regarding the recent experimental reports [Yoshikawa and Arimoto, Proc. of ISMA 2001, pp. 309–312]. The simulation results are presented in terms of the mode transient characteristics and the spectral characteristics of the synthesized sounds. They indicate that the NADM is not sufficient to describe the realistic mode transient of the recorder, while the synthesized sounds maintained almost resemble timbre to the recorder sounds.

TUESDAY AFTERNOON, 29 APRIL 2003 ROOMS 103/104, 1:00 TO 5:15 P.M.

Session 2pAO

Acoustical Oceanography, Underwater Acoustics and Signal Processing in Acoustics: Geoacoustic Inversion III

Peter Gerstoft, Chair

Scripps Institution of Oceanography, Marine Physical Laboratory, University of California, San Diego, 9500 Gillman Drive, La Jolla, California 92093-0238

Invited Paper

1:00

2pAO1. Developments in self-noise towed-array inversion. David J. Battle (Marine Physical Lab, Univ. of California, San Diego, La Jolla, CA 92093-0238, davidb@mpl.ucsd.edu)

In preliminary experiments, self-noise geoacoustic inversion from towed-array data has been demonstrated as a feasible concept, whereby sonar performance predictions in shallow water can be based on locally measured geoacoustic parameters while a tow-ship is underway. In this approach, replica fields generated using a near-field propagation model are compared to measured source fields via a conventional matched-field processor (MFP). With parameter selection controlled by a combination of local and global search algorithms, MFP output power is used as a feedback signal to optimize the match between the modeled and actual acoustic environments. In addition to the desired geoacoustic parameters, the search space includes nuisance parameters such as unknown or poorly known depths, ranges and array perturbations. This paper discusses further results and practicalities of self-noise inversion using towed-arrays, including aspects of environmental sensitivity, propagation model selection, source signature acquisition and imaging.

Contributed Papers

1:20

2pAO2. Rapid geoacoustic inversion with a curved horizontal array. Laurie T. Fialkowski, Dalcio K. Dacol, Joseph F. Lingevitch, and Elisabeth Kim (Naval Res. Lab., Washington, DC 20375)

Real-time geoacoustic inversions with a towed array are of interest for rapidly characterizing the sediment properties over changing regions, and require an efficient and accurate forward propagation model. A wavenumber integration solution using the computationally efficient method of undetermined coefficients and the Fast-Field Program is implemented with a simulated annealing optimization method and a coordinate rotation technique. The source function of interest may be broadband ship self-noise or a controlled source. The parameters sought in the inversion include, but are not limited to, geoacoustic sediment properties and receiver array geometry; the water column is assumed known. The forward propagation method is efficient and accurate for very short ranges. Results are presented of simulations with a controlled source, as well as preliminary MAPEX2000 ship self-noise. [Work supported by ONR.]

1:35

2pAO3. Forward modeling requirement for short-range geoacoustic inversion on a towed array. T. C. Yang and K. Yoo (Naval Res. Lab., Washington, DC 20375)

For geoacoustic inversion using the tow-ship noise received on a towed horizontal line array (HLA), one of the issues is speed and accuracy of the forward model. Is the normal mode good enough or is ray propagation adequate for modeling short-range forward propagation? Earlier work [Kuperman *et al.*, IEEE J. Oceanic Eng. **10** (1985)] showed that the continuum contribution arrives at beams far away from the forward directions. For practical applications, such contributions are often neglected to minimize interference from undesired sources. We examine in this paper the same issue (the importance of high angle arrivals at short range) in both the time and beam domain in the context of several bottom models (templates). Note that the high angle arrival contributions depend on the bottom type, the source–receiver range, the towed array aperture/spacing and the acoustic frequencies. The results of this analysis will shed light on the requirement of forward models at short ranges on a HLA. [Work supported by ONR.]

1:50

2pAO4. Sensitivity to array tilt and bow for broadband geoacoustic inversion using a towed array. K. Yoo, T. C. Yang, L. Fialkowski, D. Dacol, John Perkins (Naval Res. Lab., Washington, DC 20375), M. Fallat, P. Nielsen (SACLANT Undersea Res. Ctr., Italy), and M. Siderius (SAIC, La Jolla, CA 92037)

During the MAPEX 2000 experiment, broadband acoustic data from an acoustic source were received on a towed array, both towed by the same ship with the source–receiver range kept nearly constant. The data were used to invert for the geoacoustic properties of the bottom along the ship track. The experiment was conducted by the SACLANTCEN in the Sicily Strait where the bottom layer profile was also surveyed using standard seismic methods. The geoacoustic inversion results from the towed array data showed good agreement with the seismic data [M. Siderius *et al.*, J. Acoust. Soc. Am. **112**, 1523 (2002); M. Fallat *et al.*, SACLANTCEN Report No. SM-402, 2002] using a matched field correlation cost function that summed over frequencies coherently (for each phone) and summed over the phones incoherently. The authors reported that an approach that sums the matched-field correlation coherently over phones and incoherently over frequencies did not yield as good a result as the method mentioned above. The array tilt and bow may be one factor that impacts correlation functions in different ways. We study in this paper the matched field inversion sensitivity to the array tilt and bow as applied to the MAPEX 2000 data. [Work supported by ONR.]

2:05

2pAO5. Geoacoustic characterization of a range-dependent environment using towed array data. Mark Fallat, Peter Nielsen (SACLANT Undersea Res. Ctr., Viale S. Bartolomeo 400, 19138 La Spezia, Italy, fallat@saclantc.nato.int), and Martin Siderius (Sci. Applications Intl. Corp., La Jolla, CA 92037)

This paper describes geoacoustic characterization of a range-dependent environment using towed horizontal array data. Data from multiple points along a track are inverted using a short-range, range-independent scheme and the results are combined to produce a range-dependent model. Data from the MAPEX 2000 experiment, conducted by SACLANT Centre in the Mediterranean Sea, are used to determine seabed properties for a range-dependent environment. Inversion results are compared for both normal-mode and ray theory forward models. The layering structure of the range-dependent model compares favorably with a high-resolution seismic profile.

2:20–2:35 Break

2:35

2pAO6. Geoacoustic inversion and source localization from multisource broadband HLA data. Tracianne B. Neilsen (Appl. Res. Lab., Univ. of Texas, P.O. Box 8029, Austin, TX 78713-8029, neilsen@arlut.utexas.edu), Craig S. MacInnes (Pontificia Universidad Catolica del Peru, Lima, Peru), and David P. Knobles (Univ. of Texas, Austin, TX 78713-8029)

In underwater acoustics research, source localization efforts are often hampered by incorrect environmental information, and geoacoustic inversion results can be limited by errors in the source description. In addition, any data set can be contaminated by sound from other acoustic sources in the ocean. To overcome these problems, an iterative rotated coordinates inversion method [T. B. Neilsen, ''An iterative implementation of rotated coordinates for inverse problems,'' J. Acoust. Soc. Am. (submitted)] is employed in conjunction with a broadband bearing estimator [J. Krolik and D. Swingler, IEEE Trans. Acoust., Speech and Signal Process. **37**, (1989)] and matrix filters [C. S. MacInnes, IEEE J. Ocean. Eng. (submitted)] to obtain estimates of the source locations and the sensitive geoacoustic parameters from multisource broadband data received on a horizontal line array (HLA). In the iterative inversion method, subsets of rotated coordinates are used to vary both the source and the sensitive environmental parameters in a series of simulated annealing inversions. The information contained in the rotated coordinates and corresponding eigenvalues determine which parameters are varied significantly in each simulated annealing inversion. [Work supported by ONR.]

2:50

2pAO7. Unified inversion using isolated moving sources. Steven A. Stotts, Robert A. Koch, Traci B. Neilsen, and Craig S. MacInnes (Appl. Res. Labs., Univ. of Texas, P.O. Box 8029, Austin, TX 78713-8029, stotts@arlut.utexas.edu)

Geoacoustic inversion using beamformed data from a ship of opportunity has been demonstrated with a bottom mounted array [Koch and Knobles, J. Acoust. Soc. Am. **112**, 2282 (2002)]. An alternative approach transforms element level data into ''beam space,'' applies a bearing filter [MacInnes, IEEE J. Ocean. Eng. (submitted)] and transforms back to element level data prior to performing inversions [T. B. Neilsen and C. S. MacInnes (unpublished]. Automation of this filtering approach is facilitated for broadband applications by restricting the inverse transform back to element data corresponding to the degrees of freedom of the array, i.e., the effective number of elements. Examples that demonstrate this filtering technique with simulated data are presented along with comparisons to inversion results using beamformed data. Vertical and horizontal ambiguity surfaces are compared for the two approaches. Examinations of cost functions calculated within a simulated annealing algorithm reveal the efficacy of the approach. Filter performance with real data will be discussed.

3:05

2pAO8. Geoacoustic inversion by using the broadband signals. Zhenglin Li, Renhe Zhang, Jianjun Liu, Fenghua Li, and Ling Xiao (Inst. of Acoust., Chinese Acad. of Sci., Beijing 100080, PROC, lzhl@fared.ioa.ac.cn)

Three different methods are used to invert the sea bottom parameters (sound speed, density, and attenuation) based on the fact that each parameter has different effect on the sound fields. First, the vertical reflection coefficients of sea bottom and the bottom acoustic impedance are inverted from the light-bulb signals. Second, the matched-field processing method is used to the propagation bomb signal to split the density and sound speed from the impedance with an assumed attenuation. In the end, the attenuation coefficients are gotten from the transmission loss data at the different frequency band. [Work supported by the National Natural Science Foundation of China.]

3:20

2pAO9. Inversion of range-dependent geoacoustic properties in South China Sea ASIAEx01 experimental site. Altan Turgut, Bruce Pasewark, Marshall Orr (Naval Res. Lab., Acoust. Div., Washington, DC 20375), James Lynch (Woods Hole Oceanogr. Inst., Woods Hole, MA 02543), and Ching Sang Chiu (Naval Postgraduate School, Monterey, CA 93943)

Matched-field inversion of range-dependent geoacoustic properties is studied using broadband (50–200 Hz, 240–260 Hz, and 550–600 Hz) acoustic data collected in the South China Sea during the ASIAEx01 experiment. Range-dependent sediment sound-speed and attenuation profiles are inverted using a global optimization scheme based on genetic algorithms to minimize an objective function defined by the Bartlett processor output. For the forward model, an efficient coupled normal mode model is used to calculate broadband acoustic fields incorporating range-dependent bathymetry and sediment layers thickness obtained by chirp sonar surveys. Inversions were performed starting from a range-independent region with a relatively short source/receiver distance. Additional geoacoustic profiles were inverted by incorporating the previously inverted profiles as the source/receiver distance was increased. The results obtained at three different frequency bands are in good agreement, especially when the range-dependency of bathymetry and sediment layers thickness is included in the inversion. [Work supported by ONR.]

3:35

2pAO10. Geoacoustic inversion for elastic bottoms: Matching model predictions and backscatter data from littoral limestone. Robert F. Gragg, Raymond J. Soukup, and Roger C. Gauss (Naval Res. Lab., Code 7140, Washington, DC 20375)

The scattering strength of the ocean bottom as a function of angle and frequency is of fundamental importance in predicting the performance of active sonar systems, particularly in littoral waters. In this work, we apply an elastic-bottom scattering model [the small-slope model of Gragg *et al.*, ''Rough interface scattering,'' J. Acoust. Soc. Am. **110** (2001)] to the problem of matching 2–3.5 kHz acoustic backscatter from rough limestone sea floors off the Carolina Coast and in the Straits of Sicily. We use a simplex/annealing algorithm to determine model parameters that optimally fit the data. Analysis of an ensemble of such inversion runs addresses the following questions. How well can the elastic theory match data measured at sea? How sensitive is the theoretical prediction to the values of its inputs-frequency and the (seven) parameters that characterize the geoacoustics of the bottom material and the roughness of the surface? Results to date indicate more sensitivity to roughness than to geoacoustic parameters. This supports the importance of estimating in situ bottom roughness and the feasibility of using a physics-based model for that purpose. [Work supported by ONR.]

3:50

2pAO11. Time-reversal using ambient noise as a probe source. Philippe Roux, Hee Chun Song, and W. A. Kuperman (Marine Physical Lab., Scripps Inst. of Oceanogr., Univ. of California San Diego, La Jolla, CA 92093)

The Green's function between receivers in the ocean can be obtained from the temporal cross correlation of ambient noise measured simultaneously at those receivers [Ph. Roux, S. Lynch, W. A. Kuperman, and the NPAL Group, J. Acoust. Soc. Am. **112**, 2421 (2002)]. Constructing the Green's function similarly between a point receiver and an (source/receive) array from ambient noise creates, in effect, a surrogate probe source that provides data on a time reversal mirror. We show with theory and simulation that this noise-based time-reversal procedure results in a focus at the surrogate probe source. Finally, data are used to illustrate the feasibility of this process.

4:05–4:15 Break

4:15–5:15

Panel Discussion

TUESDAY AFTERNOON, 29 APRIL 2003 — ROOMS 110/111, 1:00 TO 2:15 P.M.

Session 2pEA

Engineering Acoustics: Microphones and Sources

Stephen C. Thompson, Chair

Knowles Electronics, Inc., 1151 Maplewood Drive, Itasca, Illinois 60143

Contributed Papers

1:00

2pEA1. Beamforming beyond the λ/2 limit with microphone arrays. Philippe Moquin, Stéphane Dedieu (Mitel Networks, 350 Legget Dr., Kanata, ON K2K 1X3, Canada), and Rafik A. Goubran (Carleton Univ., Ottawa, ON K15 5B6, Canada)

One limitation of microphone arrays is that the inter-microphone spacing is restricted to λ/2 of the shortest wavelength (highest frequency) of interest. For an increase in frequency range, the array must either be made smaller (thereby losing low-frequency directivity) or the number of microphones must be increased (thereby increasing cost). The other problem is that the beamwidth decreases with increasing frequency and sidelobes become more problematic. This results in significant off-axis ''coloration'' of the signals. The extension of the working frequency range for an existing narrow-band (300–3 kHz) telephony microphone array to wide-band telephony (up to 7 kHz), without modifying its geometry and the number of microphones will be shown. Microphones are embedded in a diffraction structure that provides the desired directivity at high frequencies. To provide the desired directionality at lower frequencies, beamforming of the microphones is performed using digital signal processing techniques. The combination of beamforming and embedding the microphones in a diffraction structure that provides the desired directivity at high frequencies ad-

dresses the two weaknesses that arise in previous approaches: low-frequency directivity with small arrays and high-frequency difficulties that arise in conventional sensor arrays. [Work supported in part by Carleton University.]

1:15

2pEA2. Practical considerations for a second-order directional hearing aid microphone system. Stephen C. Thompson (Knowles Electron., LLC, 1151 Maplewood Dr., Itasca, IL 60143, steve.thompson@knowles.com)

First-order directional microphone systems for hearing aids have been available for several years. Such a system uses two microphones and has a theoretical maximum free-field directivity index (DI) of 6.0 dB. A second-order microphone system using three microphones could provide a theoretical increase in free-field DI to 9.5 dB. These theoretical maximum DI values assume that the microphones have exactly matched sensitivities at all frequencies of interest. In practice, the individual microphones in the hearing aid always have slightly different sensitivities. For the small microphone separation necessary to fit in a hearing aid, these sensitivity matching errors degrade the directivity from the theoretical values, especially at low frequencies. This paper shows that, for first-order systems the directivity degradation due to sensitivity errors is relatively small. However, for second-order systems with practical microphone sensitivity matching specifications, the directivity degradation below 1 kHz is not tolerable. A hybrid order directive system is proposed that uses first-order processing at low frequencies and second-order directive processing at higher frequencies. This hybrid system is suggested as an alternative that could provide improved directivity index in the frequency regions that are important to speech intelligibility.

1:30

2pEA3. Microphone matching for hybrid-order directional arrays in hearing aid applications. Daniel M. Warren and Steve C. Thompson (Knowles Electron., LLC, 1151 Maplewood Dr., Itasca, IL 60143, daniel.warren@knowles.com)

The ability of a hearing aid user to distinguish a single speech source amidst general background noise (for example, dinner table or cocktail party conversation) may be improved by a directional array of microphones in the hearing instrument. The theoretical maximum directivity index (DI) of a first-order pairing of microphones is 6 dB, and a second-order array of three microphones is 9.5 dB, assuming all three microphones have identical frequency responses. The close spacing of microphone ports in a hearing aid body means that directivity degrades rapidly with differences in microphone sensitivities. A hybrid of first- and second-order arrays can mitigate this effect, although close microphone matching is still necessary for high directivity. This paper explores the effect of microphone mismatch on the directivity of such arrays, and describes practical criteria for selecting matched microphones out of production batches to maximize a speech intelligibility weighted directivity index. [Work supported by Knowles Electronics, LLC.]

1:45

2pEA4. Calculation of uncertainty in calibration of microphones by the pressure reciprocity technique. Peter Hanes, Lixue Wu, Won-Suk Ohm, and George S. K. Wong (Inst. for Natl. Measurement Standards, Natl. Res. Council, Ottawa, ON K1A 0R6, Canada, peter.hanes@nrc-cnrc.gc.ca)

At the primary level, acoustical measurement standards are realized through calibration of the sensitivity level of Laboratory Standard microphones by the reciprocity technique. The technique is described in International Standard IEC 61094-2, which allows for various implementations of the measurement method. The pressure sensitivity levels of a set of three microphones are determined from the electrical and acoustical transfer impedances of pairs of the microphones. The transfer impedances in turn depend on the design and performance of the measurement apparatus, the dimensions and acoustical properties of the microphones and the cavity that acts as an acoustical coupler between the microphones, and the prevailing environmental conditions. The uncertainty in the pressure sensitivity level depends on the uncertainties in these input quantities and on how the sensitivity level varies with changes in the input quantities. The ISO/IEC Guide Express: *1995 Guide to the Expression of Uncertainties in Measurement* provides internationally agreed models and guidance for evaluating the expanded uncertainty of a measurement. The uncertainty model, the nature of the input variables, and the steps involved in the calculation of the expanded uncertainty are described for the realization of a particular implementation of the reciprocity technique.

2:00

2pEA5. High-frequency monopole sound source for anechoic chamber qualification. Patrick Saussus and Kenneth A. Cunefare (Georgia Inst. of Technol., 771 Ferst Dr., Atlanta, GA 30332, gte036z@prism.gatech.edu)

Anechoic chamber qualification procedures require the use of an omnidirectional monopole sound source. Required characteristics for these monopole sources are explicitly listed in ISO 3745. Building a high-frequency monopole source that meets these characteristics has proved difficult due to the size limitations imposed by small wavelengths at high frequency. A prototype design developed for use in hemianechoic chambers employs telescoping tubes, which act as an inverse horn. This same design can be used in anechoic chambers, with minor adaptations. A series of gradually decreasing brass telescoping tubes is attached to the throat of a well-insulated high-frequency compression driver. Therefore, all of the sound emitted from the driver travels through the horn and exits through an opening of approximately 2.5 mm. Directivity test data show that this design meets all of the requirements set forth by ISO 3745.

Session 2pED

Education in Acoustics, Psychological and Physiological Acoustics, Noise, Speech Communication and ASA Committee on Standards: An Educated Consumer's Guide to Hearing Loss and Hearing Aids

Amy M. Donahue, Chair

NIDCD/NIH, Division of Human Communication, 6120 Executive Boulevard, Rockville, Maryland 20852

Chair's Introduction—1:00

Invited Papers

1:05

2pED1. Hearing and hearing loss: Causes, effects, and treatments. Richard A. Schmiedt (Dept. of Otolaryngol. and Head-Neck Surgery, Medical Univ. of South Carolina, P.O. Box 250550, Charleston, SC 29425, schmiera@musc.edu)

Hearing loss can have multiple causes. The outer and middle ears are conductive pathways for acoustic energy to the inner ear (cochlea) and help shape our spectral sensitivity. Conductive hearing loss is mechanical in nature such that the energy transfer to the cochlea is impeded, often from eardrum perforations or middle ear fluid buildup. Beyond the middle ear, the cochlea comprises three interdependent systems necessary for normal hearing. The first is that of basilar-membrane micromechanics including the outer hair cells. This system forms the basis of the cochlear amplifier and is the most vulnerable to noise and drug exposure. The second system comprises the ion pumps in the lateral wall tissues of the cochlea. These highly metabolic cells provide energy to the cochlear amplifier in the form of electrochemical potentials. This second system is particularly vulnerable to the effects of aging. The third system comprises the inner hair cells and their associated sensory nerve fibers. This system is the transduction stage, changing mechanical vibrations to nerve impulses. New treatments for hearing loss are on the horizon; however, at present the best strategy is avoidance of cochlear trauma and the proper use of hearing aids. [Work supported by NIA and MUSC.]

1:30

2pED2. Hearing loss and the central auditory system: Implications for hearing aids. Robert D. Frisina (Otolaryngol. Div., Univ. of Rochester Med. School, 601 Elmwood Ave., Rochester, NY 14642-8629, rdf@q.ent.rochester.edu)

Hearing loss can result from disorders or damage to the ear (peripheral auditory system) or the brain (central auditory system). Here, the basic structure and function of the central auditory system will be highlighted as relevant to cases of permanent hearing loss where assistive devices (hearing aids) are called for. The parts of the brain used for hearing are altered in two basic ways in instances of hearing loss: (1) Damage to the ear can reduce the number and nature of input channels that the brainstem receives from the ear, causing plasticity of the central auditory system. This plasticity may partially compensate for the peripheral loss, or add new abnormalities such as distorted speech processing or tinnitus. (2) In some situations, damage to the brain can occur independently of the ear, as may occur in cases of head trauma, tumors or aging. Implications of deficits to the central auditory system for speech perception in noise, hearing aid use and future innovative circuit designs will be provided to set the stage for subsequent presentations in this special educational session. [Work supported by NIA-NIH Grant P01 AG09524 and the International Center for Hearing & Speech Research, Rochester, NY.]

1:55

2pED3. Introduction to auditory perception in listeners with hearing losses. Mary Florentine (Inst. of Hearing, Speech, & Lang. and SLPA Dept. (151A FR), Northeastern Univ., 360 Huntington Ave., Boston, MA 02115-5000, florentin@neu.edu) and Søren Buus (Inst. of Hear., Speech, & Lang. and Commun. & Dig. Sig. Proc. Ctr., Northeastern Univ., Boston, MA 02115-5000)

Listeners with hearing losses cannot hear low-level sounds. In addition, they often complain that audible sounds do not have a comfortable loudness, lack clarity, and are difficult to hear in the presence of other sounds. In particular, they have difficulty understanding speech in background noise. The mechanisms underlying these complaints are not completely understood, but hearing losses are known to alter many aspects of auditory processing. This presentation highlights alterations in audibility, loudness, pitch, spectral and temporal processes, and binaural hearing that may result from hearing losses. The changes in these auditory processes can vary widely across individuals with seemingly similar amounts of hearing loss. For example, two listeners with nearly identical thresholds can differ in their ability to process spectral and temporal features of sounds. Such individual differences make rehabilitation of hearing losses complex. [Work supported by NIH/NIDCD.]

2:20–2:35 **Break**

2:35

2pED4. An introduction to hearing aids. Ole Dyrlund (GNReSound, Maarkaervej 2A, DK-2630 Taastrup, Denmark, odyrlund@gnresound.dk)

This presentation reviews hearing-aid development from analog to advanced digital technology. A basic hearing aid consists of a microphone, an amplification circuit that provides a gain that varies with frequency to accommodate variations in hearing loss with frequency, and a small earphone. In recent years, hearing aid technology has developed rapidly. Digital hearing aids have become commonplace and their share of the marketplace is increasing rapidly. Therefore, the main focus of this talk is signal-processing schemes in advanced digital hearing aids, including microphones with digitally controlled directional characteristics, wide-dynamic-range compression in multiple channels that allow the compression characteristics to vary with frequency, noise reduction, and feedback cancellation. Each of these signal-processing functions help address the needs of individuals with hearing losses.

3:00

2pED5. Hearing aids: Do they help and, if so, how does one know? Larry E. Humes (Dept. of Speech & Hearing Sci., Indiana Univ., Bloomington, IN 47405-7002, humes@indiana.edu)

For those individuals with sensorineural hearing loss, ranging from mild to severe in degree, the conventional hearing aid is the most appropriate rehabilitative device available. Despite the fact that such devices have been available commercially for over 60 years, until recently, relatively little research has been directed at evaluating the effectiveness of these rehabilitative devices. How does one evaluate the effectiveness of a hearing aid as a rehabilitative device? Should effectiveness be based on the relative improvement in communication with and without the hearing aid, typically referred to as hearing-aid benefit, the satisfaction of the consumer with the device, or simply whether and how much the hearing aid is used? How are these aspects of hearing-aid effectiveness or outcome measured? Are the measures of hearing-aid outcome related to one another? What evidence is there regarding the effectiveness of contemporary hearing aids? Recent research regarding these and other related questions will be reviewed in this presentation. [Work supported, in part, by NIA.]

3:25

2pED6. Beyond the hearing aid: Assistive listening devices. Alice E. Holmes (Dept. of Communicative Disord., Univ. of Florida, P.O. Box 100174, Gainesville, FL 32610, aholmes@hp.ufl.edu)

Persons with hearing loss can obtain great benefit from hearing aids but there are many situations that traditional amplification devices will not provide enough help to ensure optimal communication. Assistive listening and signaling devices are designed to improve the communication of the hearing impaired in instances where traditional hearing aids are not sufficient. These devices are designed to help with problems created by listening in noise or against a competing message, improve distance listening, facilitate group conversation (help with problems created by rapidly changing speakers), and allow independence from friends and family. With the passage of the Americans with Disabilities Act in 1990, assistive listening devices (ALDs) are becoming more accessible to the public with hearing loss. Employers and public facilities must provide auxiliary aids and services when necessary to ensure effective communication for persons who are deaf or hard of hearing. However many professionals and persons with hearing loss are unaware of the various types and availability of ALDs. An overview of ALDs along with a discussion of their advantages and disadvantages will be given.

Contributed Paper

3:30

2pED7. Teaching hearing science to undergraduate nonscientists. Ernest M. Weiler (CSD, ML #394, College of Allied Health Sci., Univ. of Cincinnati, Cincinnati, OH 45267-0394, ernest.weiler@uc.edu), Suzanne Boyce, and Joseph Steger (Univ. of Cincinnati, Cincinnati, OH)

For those students interested in potential clinical careers in Speech Pathology, or Audiology, a knowledge of some of the scientific bases is important, but should not create a distaste for science. The authors have addressed themselves to these goals: (1) calculation of period, Hz, summation of two sine waves, phase and dB; (2) anticipating undergraduate Speech Science; (3) simple examples of hearing pathology; and (4) basic psycho-acoustical issues. The classic material of Harry Helson was used to elucidate issues of context in experimental science, and that of S.S. Stevens was used to exemplify psycho-acoustical formulas of common use. Four texts that have been tried on approximately 200 students were evaluated. Surprisingly, the best provided the fewest formulas, short study questions with answers, good examples, and a list of common terms. The next best was aimed at slightly more advanced students, but each chapter contained introductory material, examples, and definitions suitable for naïve undergraduates. The least satisfactory text provided excerpts of technical material with abrupt transitions, no examples, and only part of the definitions needed for the naïve student. Perhaps the most difficult teaching issue is to avoid demanding graduate-level science from those undergraduates with clinical aspirations.

Session 2pPA

Physical Acoustics: Sono(con)-fusion II: Evaluating the Chances and Claims of Bubble Fusion

D. Felipe Gaitan, Cochair
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R. Glynn Holt, Cochair
Aerospace and Mechanical Engineering, Boston University, 110 Cummington Street, Boston, Massachusetts 02215

Thomas J. Matula, Cochair
Applied Physics Laboratory, University of Washington, 1013 N.E. 40th Street, Seattle, Washington 98105

Invited Papers

1:45

2pPA1. Sonoluminescence and multi-bubble cavitation phenomena for selected research and industrial applications. Larry Greenwood, Khris Olsen, Morris Good, Leonard Bond (PNNL, P.O. Box 999, P7-22, Richland, WA 99352), Gerald Posakony, Timothy Peters, David Baldwin, Dennis Wester, and Salahuddin Ahmed (PNNL, Richland, WA 99352)

Single bubble sonoluminescence (SBSL), multi-bubble sonoluminescence (MBSL), multi-bubble sonochemiluminescence (MB-SCL) and other high power ultrasound cavitation and noncavitating ultrasound process stream interaction phenomena are known to produce a wide range of both physical and chemical effects that depend upon the system and operating conditions employed. Three interacting regimes are under investigation (a) high power and high frequency (including noncavitating systems), (b) single bubble resonance/sonoluminescence and (c) multi-bubble high power sonochemical processing. In all cases these involve various reactors, including possible schemes for continuous material feeding and processing for selected chemical, nonaqueous fluids and biological research and industrial applications. High power sonochemical and noncavitating ultrasound processing applications and a review of literature pertaining to the potential of high power processing, including fusion are discussed. Work includes the investigation of acoustic fields in reactors, characterization of sonoluminescence spectra, the investigation of system parameters to control maximum bubble temperature and pressure, and acoustic energy partition into light and acoustic emission/shock waves. Effects of various chemical systems on multi-bubble luminescence are being investigated and will be reported. Work to date has emphasized the evaluation of both single and multi-bubble sonoluminescence, spectral measurements, acoustic emission measurements and the observation of a continuous bubble feed phenomenon.

2:00

2pPA2. Basic physics boundary conditions of acoustically driven, inertial confinement fusion. Lawrence Forsley (JWK Intl. Corp., Ste. 800, 7617 Little River Turnpike, Annandale, VA 22003, lforsley@jwk.com), Robert August, Robert Whitlock (Naval Res. Lab., Washington, DC), and Jacques Deletrez (LLE, Rochester, NY 14623)

This paper defines boundary conditions derived from the basic physics of fusion and applies them to both laser driven and acoustically driven Inertial Confinement Fusion (ICF). Several experimental and theoretical papers, in addition to some patents, hold open the promise of acoustically driven ICF. There are several factors common to both drivers that must be taken into account. In particular, it has been observed in laser driven ICF plasmas that criteria on the ion temperature, the confinement time, the core density, and the minimum core radius must be satisfied to achieve fusion. The relationship of these criteria to acoustically driven inertial confinement will be discussed.

2:25

2pPA3. Observing sonoluminescent UV photons below the water cutoff. Robert August (NRL, Washington, DC 20375-5321, robert.august@nrl.navy.mil), Lawrence Forsley (JWK Intl. Corp., Annandale, VA 22003), and Robert Whitlock (NRL, Washington, DC 20375-5321)

This paper presents sonoluminescent UV photon data observed in water below the water cut-off (200 nm) with hermetically sealed VUV photodiodes. Carbon and thin film metal filters provide both the encapsulation against water as well as the band-pass filtering of the diodes allowing spectra to be inferred. No previous data exists in this spectral region due to an effectively infinite absorption of photons after a few microns of water. The difficulty of these measurements is amplified by the requirement to use a detector normally intended for use in a high vacuum system. These measurements are important because the spectral shape in this region is unknown. We expect from published measurements at longer wavelengths that the photon fluence in this region should be high. These measurements may allow a differentiation between several competing sonoluminesence theories.

2:45

2pPA4. Update and clarifications on experimental studies for nuclear emissions during acoustic cavitation. Rusi P. Taleyarkhan, C. D. West, J. S. Cho (Oak Ridge Natl. Lab., Oak Ridge, TN 37831), R. T. Lahey, R. C. Block (Rensselaer Polytechnic Inst., Troy, NY), and R. I. Nigmatulin (Russian Acad. of Sci., Russia)

A seminal discovery related to detection of nuclear emissions during acoustic inertial confinement fusion with deuterated acetone has been reported in *Science* (3/8/2002 issue). Nuclear emissions we measured included 2.5-MeV neutrons and tritium as would be expected from deuterium–deuterium nuclear fusion. These unmistakable statistically significant signatures were measured under conditions commensurate with degassed rapid condensation-induced implosion conditions only with the test fluid deuterated acetone. In these experiments bubble clusters are nucleated in tensioned degassed liquids with neutrons at the nanoscale level and are then made to grow by a factor of ~100 000 in size to the mm scale prior to implosive collapse. Similarly conducted control experiments with natural acetone did not result in any statistically significant nuclear emissions. Shock code simulations (discussed in a companion talk) corroborated these observations and provided insights into the physics of the overall process. Since the recent announcement of this discovery several world-wide researchers have contacted the authors for further clarifications in a variety of areas. The presentation will discuss these issues and questions, and will provide relevant explanations with supporting evidence.

3:30

2pPA5. Nuclear fusion in collapsing bubbles—Is it there? An attempt to repeat an experiment that reported d–d fusion in bubble collapse induced by cavitation in deuterated acetone. Dan Shapira and Mike Saltmarsh (Phys. Div., Oak Ridge Natl. Lab., P.O. Box 2008, M.S. 6368, Oak Ridge, TN 37831-6368)

The experiment of Taleyarkhan *et al.* [Science **295**, 1686 (2002)] has been repeated [D. Shapira and M. Saltmarsh, Phys. Rev. Lett. **89**, 104302 (2002)] in an attempt to detect the emission of neutrons from d–d fusion during bubble collapse in deuterated acetone. Using the same apparatus and method for bubble seeding and cavitation but a more sophisticated data acquisition system, and a large liquid scintillator detector we find no evidence for 2.5 MeV neutron emission correlated with sonoluminscence from the collapsing bubbles. Any neutron emisssion that might occur is at least four orders of magnitude smaller than that necessary to explain the tritium production reported in Taleyarkhan *et al.* as being due to d–d fusion. We demonstrate that the proper allowance for random coincidence rates in such experiments requires the simultaneous measurement of the complex time-varying singles rates.

4:15–4:30 Break

4:30–5:30

Panel Discussion

TUESDAY AFTERNOON, 29 APRIL 2003 ROOM 206, 1:00 TO 4:00 P.M.

Session 2pPP

Psychological and Physiological Acoustics: Pitch, Temporal Effects and Hearing Impairment (Poster Session)

Jennifer Lentz, Chair

Department of Speech and Hearing Science, Indiana University, 200 South Jordan, Bloomington, Indiana 47405

Contributed Papers

All posters will be on display from 8:00 a.m. to 4:00 p.m. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 1:00 p.m. to 2:30 p.m. and contributors of even-numbered papers will be at their posters from 2:30 p.m. to 4:00 p.m. To allow for extended viewing time, posters will be on display beginning at 8:00 a.m.

2pPP1. Simulations of cochlear implant hearing using filtered harmonic complexes: Implications for concurrent sound segregation and pitch perception. John M. Deeks and Robert P. Carlyon (MRC Cognition & Brain Sci. Unit, 15 Chaucer Rd., Cambridge CB2 2EF, UK)

We studied concurrent sound segregation using cues similar to those available to cochlear implant listeners. Sixteen normally hearing subjects listened to mixtures of IHR sentences, processed using a simulation of implant hearing. Target sentences were bandpass filtered into six frequency bands between 1020 and 5000 Hz. The amplitude envelope in each band modulated a filtered, alternating-phase harmonic complex with $F0 = 40$ or 70 Hz. Each complex resembled a pulse train with pulse rate $= 2F0$ and was filtered in the same way as the speech band that modulated it. The complexes consisted only of unresolved harmonics, whose pitch is processed in a way similar to that of electric pulse trains in implant hearing. The interferer was 3.5 s of time-reversed speech, processed in the same way as the target. After processing, targets were added to interferers at a 1.2-s delay, with SNR=9 dB. The target and interferer used complexes with either the same or different $F0$. Using a different $F0$ for the interferer benefited performance when the target $F0=70$ Hz, but had no effect when the target $F0=40$ Hz. Modifying the scheme by processing one sentence on only the odd-numbered channels and the other on the even-numbered channels impaired performance in all conditions.

2pPP2. The effects of simulated cochlear-implant processing on $F0$ discrimination. Michael K. Qin and Andrew J. Oxenham (MIT Res. Lab. of Electron. and Harvard-MIT Div. of Health Sci. and Technol., SHBT Prog., 3 Ames St., Cambridge, MA 02139-4307)

Fundamental frequency ($F0$) information has long been thought to play an important role in perceptually segregating simultaneous and nonsimultaneous sources. The ability to discriminate different $F0$'s is thought to depend primarily on fine-structure information, in particular the information carried in peripherally resolved, lower-order harmonics. Current cochlear-implant users do not have access to such cues. While implant-processed stimuli can carry some periodicity information in the stimulus envelopes, the F0 discriminability associated with envelope periodicity is rather weak. This study investigated $F0$ discrimination of complexes with $F0$'s of 130 and 220 Hz in normal-hearing listeners using noise-excited vocoders to simulate cochlear-implant processing. $F0$ difference limens of tone complexes were measured as a function of processing condition (1, 4, 8, 24, 40 channels, and unprocessed) and environment (tone complex alone and in reverberation). Performance with vocoder processing was poorer than in unprocessed conditions, despite the relatively good frequency resolution with the highest number of channels. Vocoder processing was particularly detrimental to $F0$ discrimination in the reverberant conditions. The detrimental effects of vocoder processing may be due to the elimination of temporal fine-structure information or the poorer spectral representation of the lower-order harmonics. [Work supported by NIDCD Grant No. R01 DC05216.]

2pPP3. Viability of spectral enhancement with harmonic stimuli. Jeffrey J. DiGiovanni (W235 Grover Ctr., School of Hearing Speech and Lang. Sci., Ohio Univ., Athens, OH 45701)

Loss of spectral resolution is an established consequence of sensorineural hearing loss. Traditional hearing aid design includes amplification and compression. These do not, however, account for the loss in frequency resolution. Recently, spectral enhancement processing has been designed to at least partially restore aspects of frequency resolution. The critical feature of this design is to increase the peak to trough ratio of the speech spectrum. These have been implemented with mixed success [e.g., Miller *et al.* (1999); Franck *et al.* (1999)]. More recently, DiGiovanni *et al.* (2002) showed promising results for normal and hearing-impaired subjects with psychophysical noise stimuli. The goal of this study was to expand these results to harmonic stimuli while adding peaks at fixed formant places within the spectrum. In that regard, subjects listened in two psychophysical experiments: detecting an $F2$-like spectral increment in a broadband harmonic complex and detecting the increment with an additional fixed formant peak added at an appropriate $F1$ place. Preliminary results show that normally hearing subjects have an improved ability to detect a narrowband tone complex when there is a spectral decrement at frequencies adjacent to the increment. These results are further support that the idea of spectral enhancement is viable.

2pPP4. Neuroimaging of speech recognition under conditions of spectral reduction and frequency upshift. C.-Y. Peter Chiu (Dept. of Psych. & Dept. of Commun. Sci. and Disord., 401A Dyer Hall, Univ. of Cincinnati, Cincinnati, OH 45229-0376, peter.chiu@uc.edu)

In the current study explored the cortical dynamics of speech recognition, given spectral reduction and frequency upshifts, using functional MRI. Subjects with normal hearing either rested or listened to speech under different conditions. In the 8-channel condition, natural speech was processed by an 8-channel sinewave vocoder to remove its fine spectral details [Shannon *et al.*, J. Acoust. Soc. Am. **104**, 2467 (1998)]. In the upshifted condition, the carrier center frequency of each of the 8 channels was further shifted upward in frequency from the corresponding analysis band by "6 mm" in cochlear frequency space [Fu and Shannon, J. Acoust. Soc. Am. **105**, 1889 (1999)]. All subjects received a brief practice session with the speech stimuli prior to scanning. In Experiment 1, subjects listened to nonmonosyllabic words and pressed a key whenever they heard a concrete noun. In Experiment 2, subjects listened to high context sentences (SPIN) and pressed a key whenever they recognized all the words in a particular sentence. Preliminary data suggested that, compared to rest, all speech conditions evoked comparable activities in largely similar sets of bilateral superior temporal regions, with relatively minor differences between words and sentences. Activation appeared to be least diffuse in the natural speech condition.

2pPP5. Learning to recognize speech that is spectrally reduced and frequency upshifted. Marie E. McCabe (Dept. of Psych., Dyer Hall, ML 0376, Univ. of Cincinnati, Cincinnati, OH 45229-0376, mariemccabe@yahoo.com) and Peter Chiu (Univ. of Cincinnati, Cincinnati, OH 45229-0376)

The current study explored to what extent training could ameliorate the deleterious effect of large frequency upshifts in spectrally reduced speech. During each training session, subjects attempted recognition of IEEE sentences spoken by a single talker once and received feedback for their responses. Training sentences were processed to simulate an 8-channel CIS cochlear implant processor with a "6 mm frequency upshift" [Fu and Shannon, J. Acoust. Soc. Am. **105**, 1889 (1999)]. Three test sessions were administered to all subjects to assess recognition of sentences (IEEE and HINT), consonants (/aCa/), and vowels (in /hVd/ and /bVt/ contexts) pre-, post-, as well as at the mid-point of training. Four processing conditions (i.e., unprocessed, 8-channel-unshifted, 8-channel-upshifted, and 8-channel-upshifted-and-compressed) were tested for each type of materials. Preliminary data suggest that performance improved for most subjects during training, but there were substantial individual differences in learning rates and asymptotic performance levels. Vowels were more difficult to recognize and showed smaller training-related gains when compared to consonants and sentences. The rank ordering of recognition performance was consistent among the four processing conditions (unshifted best; upshifted-and-compressed intermediate; upshifted worst) for all measures. Data comparing the efficacy of an alternative training method will also be presented.

2pPP6. Estimates of pitch strength for musicians and nonmusicians. Marsha G. Clarkson, Cynthia M. Zettler, Michelle J. Follmer, Margaret Faulk (Dept. of Psych., Georgia State Univ., Atlanta, GA 30303), and Michael J. Takagi (Monash Univ., Melbourne, Australia)

To measure the strength of the pitch of iterated rippled noise (IRN), 19 adults were tested in an operant conditioning procedure. Seven adults had music training and currently played an instrument; 12 adults had no training and did not currently play an instrument. To generate IRN, a 500-ms Gaussian noise stimulus was delayed by 5 or 6 ms (pitches of 200 or 166 Hz) and added to the original for 16 iterations. IRN stimuli having one delay were presented repeatedly. On signal trials the delay changed for 6 s. Stimulus level roved from 63-67 dBA (background of 28 dBA). Adults learned to press a button when the stimulus changed. Testing started with IRN stimuli having 0-dB attenuation (i.e., maximal pitch strength). Stimuli having weaker pitches (i.e., progressively greater attenuation applied to the delayed noise) followed. Strength of pitch was quantified as the maximum attenuation for which pitch was discerned. For each subject, threshold attenuation for pitch strength was extrapolated as the 71% point on a psychometric function depicting percent correct performance as a function

of attenuation. Mean thresholds revealed that the pitch percept was similar for both nonmusically trained (18.70 dB) and musically trained adults (18.73 dB).

2pPP7. Further tests of the "two pitch mechanisms" hypothesis. Christophe Micheyl and Andrew Oxenham (Res. Lab. of Electronics, MIT, Bldg. 36-797, Cambridge, MA 02139-4307)

To further investigate the hypothesis that $F0$ is encoded via different mechanisms for resolved and unresolved harmonics, we measured $F0$ difference limens (DL$F0$'s) between two groups of harmonics that were both resolved, both unresolved, or differed by resolvability (one resolved, one unresolved). In a first experiment, the complexes were filtered in the same or a different frequency region and presented sequentially; listeners had to indicate which had the higher pitch. In the second experiment, two pairs of simultaneous complexes filtered in a different region were presented sequentially; subjects had to indicate the interval containing the pair of complexes with different $F0$'s. To reduce possible pitch-pulse-asynchrony cues, the starting phase of the entire complex was randomized across spectral regions in. The results of the first experiment showed that for tracks where the higher $F0$ was in the higher spectral region (consistent pitch-timbre cues), DL$F0$'s were consistently smaller for same-resolvability than for different-resolvability complexes. On the other hand, the preliminary results of the second experiment, using simultaneous complexes, do not provide clear support for the hypothesis that DL$F0$'s for resolved versus unresolved comparisons are impeded by "translation" noise between the output of the two pitch mechanisms. [Work supported by NIDCD R01DC05216.]

2pPP8. A neural network model of harmonic detection. Clifford F. Lewis (Dept. of Psych., Kent State Univ., P.O. Box 5190, Kent, OH 44242-0001, clewis@kent.edu)

Harmonic detection theories postulate that a virtual pitch is perceived when a sufficient number of harmonics is present. The harmonics need not be consecutive, but higher harmonics contribute less than lower harmonics [J. Raatgever and F. A. Bilsen, in *Auditory Physiology and Perception*, edited by Y. Cazals, K. Horner, and L. Demany (Pergamon, Oxford, 1992), pp. 215–222; M. K. McBeath and J. F. Wayand, Abstracts of the Psychonom. Soc. **3**, 55 (1998)]. A neural network model is presented that has the potential to simulate this operation. Harmonics are first passed through a bank of rounded exponential filters with lateral inhibition. The results are used as inputs for an autoassociator neural network. The model is trained using harmonic data for symphonic musical instruments, in order to test whether it can self-organize by learning associations between co-occurring harmonics. It is shown that the trained model can complete the pattern for missing-fundamental sounds. The Performance of the model in harmonic detection will be compared with experimental results for humans.

2pPP9. Effects of modulation phase on profile analysis in normal-hearing and hearing-impaired listeners. Deanna Rogers and Jennifer Lentz (Dept. of Speech and Hearing Sci., Indiana Univ., Bloomington, IN 47408, jjlentz@indiana.edu)

The ability to discriminate between sounds with different spectral shapes in the presence of amplitude modulation was measured in normal-hearing and hearing-impaired listeners. The standard stimulus was the sum of equal-amplitude modulated tones, and the signal stimulus was generated by increasing the level of half the tones (up components) and decreasing the level of half the tones (down components). The down components had the same modulation phase, and a phase shift was applied to the up components to encourage segregation from the down tones. The same phase shift was used in both standard and signal stimuli. Profile-analysis thresholds were measured as a function of the phase shift between up and down components. The phase shifts were 0, 30, 45, 60, 90, and 180 deg. As expected, thresholds were lowest when all tones had the same modulation phase and increased somewhat with increasing phase disparity. This small increase in thresholds was similar for both groups. These results suggest that hearing-impaired listeners are able to use modulation phase to group sounds in a manner similar to that of normal listeners. [Work supported by NIH (DC 05835).]

2pPP10. Perceptual learning in frequency discrimination and amplitude-modulation rate discrimination, and generalization to fundamental frequency discrimination. Nicolas Grimault, Christophe Micheyl (UMR, CNRS 5020 UCBL1, 50 av Tony Garnier, 69366 Lyon, Cedex 07, France, nicolas.grimault@olfac.univ-lyon1.fr), Robert P. Carlyon (MRC Cognition and Brain Sci. Unit, Cambridge CB2 3EF, UK), Sid P. Bacon (Arizona State Univ., Tempe, AZ 85287-1908), and Lionel Collet (CNRS 5020 UCBL1, 69366 Lyon, Cedex 07, France)

Fifteen subjects were trained during twelve 2-h sessions in either frequency discrimination with pure tones, or amplitude-modulation rate discrimination of noise bands. Thresholds for the discrimination of pure-tone frequency, harmonic complex tone fundamental frequency, and amplitude-modulation rate were measured before, during, and after training. Comparison of pre- and post-training thresholds revealed significant improvements in all conditions in both subjects trained in frequency discrimination and subjects trained in modulation rate discrimination. Training in frequency discrimination resulted in larger improvements in fundamental frequency discrimination when the test complexes contained resolved harmonics than when they were composed of unresolved harmonics. Training in modulation rate discrimination did not result in larger fundamental frequency discrimination improvements for unresolved than for resolved harmonics. The implications for models of pitch perception are discussed.

2pPP11. The effects of three temporal cues on the detection of increments and decrements in intensity. Yang-soo Yoon and David M. Gooler (Dept. of Speech and Hearing Sci., Univ. of Illinois at Urbana–Champaign, Champaign, IL 61820, yyoon5@uiuc.edu)

This study investigated the effect of three temporal cues: interonset interval, interpulse interval, and pulse duration, on auditory intensity detection. Eight adults with normal hearing served as subjects. The level detection (target increment or decrement) ability of listeners was measured in a sequential oddball paradigm where targets occurred randomly. The experiment used spectrally identical, broadband noise bursts as standard (30 dB above threshold to a train of standard pulses) and target pulses (1, 2, and 3 dB *re*: standard pulse level). At each target level performance was measured for different pulse interonset intervals (20 ms to 180 ms), interpulse intervals (10 ms to 90 ms), and pulse duration (10 ms to 90 ms, 0.5 ms rise/fall time). Overall, the results showed that the performance of the intensity detection was more influenced by the interpulse interval than the interonset interval. Also, performance was dependent on pulse duration since a poorer performance with longer interpulse intervals could be improved by increasing pulse duration. Performance at different inter-pulse intervals may reflect different perceptual strategies for comparing acoustic events that occur in a sequence.

2pPP12. Effect of duration and inter-stimulus interval on auditory temporal order discrimination in young normal-hearing and elderly hearing-impaired listeners. Mini M. Narendran and Larry E. Humes (Dept. of Speech and Hearing Sci., Indiana Univ., 200 S. Jordan Ave., Bloomington, IN 47405, mnarendr@indiana.edu)

Increasing the rate of presentation can have a deleterious effect on auditory processing, especially among the elderly. Rate can be manipulated by changing the duration of individual components of a sequence of sounds, by changing the inter-stimulus interval (ISI) between components, or both. Consequently, when age-related deficits in performance appear to be attributable to rate of stimulus presentation, it is often the case that alternative explanations in terms of the effects of stimulus duration or ISI are also possible. In this study, the independent effects of duration and ISI on the discrimination of temporal order for four-tone sequences were investigated in a group of young normal-hearing and elderly hearing-impaired listeners. It was found that discrimination performance was driven by the rate of presentation, rather than stimulus duration or ISI alone, for both groups of listeners. The performance of the two groups of listeners differed significantly for the fastest presentation rates, but was similar for the slower rates. Slowing the rate of presentation seemed to improve performance, regardless of whether this was done by increasing stimulus duration or increasing ISI, and this was observed for both groups of listeners. [Work supported, in part, by NIA.]

2pPP13. Difference limen for perception of aspiration noise. Rahul Shrivastav and Christine M. Sapienza (Dept. of Commun. Sci. & Disord., Univ. of Florida, Dauer Hall, Gainesville, FL 32611)

The relationship between the vocal acoustic signal and the perception of voice quality has been the subject of much research. An increase in the intensity of aspiration noise is associated with a perception of greater breathiness in voices [Klatt and Klatt (1990); Childers (1993)]. Shrivastav (2001) found that subjective ratings of breathiness could be better predicted using measures calculated from an auditory spectrum of the vocal acoustic signal. These measures were found to vary with changes in the spectral slope as well as the intensity of aspiration noise in voice. The aim of the present experiment was to determine the smallest change in the intensity of aspiration noise that could be perceived by listeners. Six voice continua, with increasing intensity of aspiration noise, were generated using the Klatt synthesizer. Five listeners participated in an adaptive listening test, where these stimuli were presented in pairs and listeners were asked to identify the stimuli in each pair as being same or different. Listener responses were used to determine the difference limen for changes in intensity of aspiration noise in voice. These findings will help understand the perception of breathy voice quality and to develop objective measures for its quantification.

2pPP14. The aging middle ear: Wideband energy reflectance measurements. M. Patrick Feeney (Dept. of Otolaryngol., Head and Neck Surgery, V. M. Bloedel Hearing Res. Ctr., Univ. of Washington, Box 357923, Seattle, WA 98195) and Chris A. Sanford (Univ. of Washington, Seattle, WA 98195)

Several anatomical studies have documented aging effects in the human middle ear. However, efforts to study the effect of aging using both low-frequency and multifrequency tympanometry to 2000 Hz have been inconclusive. This study examined energy reflectance at ambient pressure from 250 to 10,080 Hz in 40 young ($M=22$ years) and 34 elderly adults ($M=72$ years). All subjects had normal 226 Hz tympanometry and audiometric air-bone gaps of 10 dB or less. Reflectance measurements were obtained in a sound-treated booth using a digitally-generated wideband chirp as the probe stimulus delivered by a receiver in an ER-10C microphone. Each reflectance measurement consisted of a time-waveform average of the microphone response to 8 chirps. Three such one-third-octave reflectance responses were averaged to obtain an estimate of middle ear reflectance for one ear of each subject. The average reflectance for the elderly subjects was significantly lower from 794 to 2000 Hz with a maximum difference of 15% at 1260 Hz. A reflectance minimum occurred at 4000 Hz for both groups, but was about 15% lower for the young group. Results will be compared to published adult data using similar systems. [Work supported by the NIDCD Grant No. DC04129.]

2pPP15. Evidence of peripheral nonlinearity in psychometric function slopes of forward-masked tones at 250 and 4000 Hz. Kim S. Schairer and Walt Jesteadt (Boys Town Natl. Res. Hospital, 555 N. 30th St., Omaha, NE 68131)

Results from this laboratory suggest that the compressive nonlinearity of the basilar membrane is reflected in slopes of psychometric functions (PFs) for forward-masked tones of 4000 Hz. Briefly, as the signal level at the threshold increases, slopes of PFs decrease. Many other behavioral measures have suggested significant nonlinearity at high frequencies, but results are less consistent at low frequencies. The purpose of the current study was to use a PF slope to investigate nonlinearity using low- and high-frequency stimuli in the same normal-hearing adults. Signals were 250 or 4000 Hz, 10-ms (5-ms rise/fall) duration, presented with a 10-ms delay. The on-frequency forward-maskers were 200-ms (2-ms rise/fall), presented at levels of 30-, 50-, 70-, and 90-dB SPL in separate conditions. A two-track, two-interval-forced-choice adaptive procedure was used, with decision rules to estimate 71% correct on one track, and 87% on the other track. Data from both tracks were combined to estimate PF slopes. PF slopes were steeper in low-threshold conditions and shallower in high-threshold conditions for both frequencies. These results suggest a significant nonlinearity at 250 Hz. [Work supported by NIDCD.]

2pPP16. Confidence ratings and awareness measures in word recognition testing. Edward L. Goshorn and Jennifer D. Goshorn (Speech Dept., P.O. Box 3165 Tech Station, Louisiana Tech Univ., Ruston, LA 71272)

Word recognition tests primarily use percent correct to measure performance. Additional information may be gained by analyzing awareness of accurate perceptions (AA), awareness of errant perceptions (AE), composite awareness (AC), and awareness symmetry (AS). Awareness measures were derived from subjects' assignment of confidence ratings to a two-item multiple choice response test by designating "YES" or "NO" that their chosen response is accurate. Each response/confidence rating was categorized as a hit, miss, false alarm, or correct rejection. Awareness equations were: AA=hits/(hits+misses); AE=correct rejections/(correct rejections+false alarms); AC=SQRT (AA squared+AE squared); AS =0.707(AA−AE). Thus, AC is the vector to Cartesian coordinates AA, AE; AS is the distance of this point from a diagonal representing symmetrical awareness. Word recognition and awareness was investigated under two signal-to-noise ratios (3 and 6 dB). The Diagnostic Rhyme Test was presented at 50 dBHL to eight normal-hearing adults. Six replicates were obtained. Awareness measures provided additional performance information. Percent correct increased significantly as signal-to-noise ratio improved, but AE decreased and AS did not change significantly.

Session 2pSA

Structural Acoustics and Vibration and Noise: Damping and Absorption

Kenneth A. Cunefare, Chair
School of Mechanical Engineering, Georgia Institute of Technology, Atlanta, Georgia 30332-0405

Contributed Papers

2:30

2pSA1. Optimization of a state-switched absorber applied to a vibrating continuous system. Mark Holdhusen and Kenneth Cunefare (The George W. Woodruff School of Mech. Eng., Georgia Inst. of Technol., Atlanta, GA 30332-0405)

A state-switched device is a device that can instantaneously change one or more of its dynamic properties. A vibration absorber that can switch between discrete stiffenesses is termed a state-switched absorber. Using a maximum work extraction switching rule, the state-switched absorber has been shown to increase performance in lumped mass systems as compared to a classical tuned vibration absorber. Because the state-switched absorber can retune its resonance frequency during the response of the system, the bandwidth which the state-switched absorber is effective is larger than that of classical passive devices. The work at hand considers the effectiveness of vibration control using state-switched absorber attached to a continuous system subjected to a multifrequency excitation. The switching rule, tuning frequencies, and location of attachment on the continuous system are optimized for the state-switched absorber to achieve the lowest kinetic energy in the insolated system. Tuning and location optimization is also done for a classical tuned vibration absorber. The optimized performance of the state-switched absorber is compared to the optimized performance of the tuned vibration absorber for a number of different forcing cases.

2:45

2pSA2. A proposed definition for the induced noise control parameter. G. Maidanik and K. J. Becker (NSWCCD (DTMB), Code 7030, 9500 MacArthur Blvd., West Bethesda, MD 20817)

The response of an externally force-driven master dynamic system is examined under two conditions. In the first, the master dynamic system is in isolation and the (quadratic) response is stated in terms of its stored energy $E_0^0(\omega)$, where (ω) is the (angular) frequency. In the second, the master dynamic system is passively coupled to an adjunct dynamic system that is not externally driven. The response of the master dynamic system, that is subjected to the same external force drive, is changed due to the couplings. The change is from $E_0^0(\omega)$, in isolation, to $E_0(\omega)$, when coupled. The induced noise control parameter $\xi(\omega)$ is then defined: $\xi(\omega)=[E_0(\omega)/E_0^0(\omega)]$. The smaller $\xi(\omega)$ is, the more benefit, by the couplings, is accrued to the noise control of the master dynamic system. When $\xi(\omega)$ exceeds unity, by definition, an induced noise control reversal occurs. Illustrations of an induced noise control are provided in terms of a master dynamic system represented by an (harmonic) oscillator and an adjunct dynamic system represented by a set of satellite (harmonic) oscillators. The satellite oscillators are not coupled to each other; they are individually coupled to the master oscillator only. A coupling may include a combination of mass, stiffness, and gyroscopic elements.

3:00

2pSA3. Effect of burst parameters on automotive brake squeal suppression. Jeff Badertscher and Kenneth A. Cunefare (Georgia Inst. of Technol., 771 Ferst Dr., Atlanta, GA 30332, gtg437d@prism.gatech.edu)

Implementing a dither control signal with a 100% duty cycle is an effective means of suppressing automotive brake squeal. Dither control is a method by which high-frequency control efforts are introduced into a system to suppress a lower frequency disturbance. Dither is introduced to a brake by placing a piezoelectric stack actuator in the piston of a floating caliper brake. Burst mode dither control is characterized by duty cycles of less than 100%. A burst control signal of a specific duty cycle is also specified by the burst count and burst rate. Burst mode signals are shown to suppress brake squeal. This paper examines the nature of suppression and the effectiveness of burst mode dither control signals with varied burst parameters. An examination of the squeal response and dither control signal is used to examine the nature of suppression during bursting and dwell time. The amplitude of the control signal that is necessary to obtain full control of the system is used to assess control signal effectiveness.

3:15

2pSA4. Modeling the effect of bearing properties on the eigenvalues of a rotordynamic system. Ben B. Wagner and Jerry H. Ginsberg (Georgia Inst. of Technol., Woodruff School of Mech. Eng., 801 Ferst Dr. NW, Atlanta, GA 30332-0405)

Experimental data indicate that changes in bearing properties alter the natural frequency and damping ratios of rotordynamic systems [R. M. Baldwin, ASME 92-GT-53]. The work presented here examines the development of a mathematical model that will be used to investigate how natural frequency, modal damping ratio, shaft rotation rate, bearing clearance, and lubricant viscosity are related. The modeled system consists of a uniform, elastic, rotating shaft, which is supported by two plain journal bearings, and a single, rigid disk, which is concentrically welded to the shaft away from midspan. Standard lubrication theory is used to generate the stiffness and damping matrices for the bearings. A Ritz series expansion is used to generate the mass, shaft stiffness, and gyroscopic matrices describing the shaft and disk. The combined action of the bearings and shaft/disk system is described by matrices representing the effects of stationary and rotational inertia, stiffness, and internal and external damping. The radial clearance and lubricant viscosity are independently adjustable at each bearing, and the range of shaft speed for analysis is user-specified. A nonsymmetric generalized eigenvalue problem solver is used to calculate eigenvalues, whose real part is proportional to the modal damping ratio, and whose imaginary part is the natural frequency. For Structural Acoustics and Vibrations Best Student Paper Award. Submitted for Structural Acoustics and Vibrations Young Presenter Award.

3:30

2pSA5. What is the best method for structural damping identification? Part 1: A survey of methods. Srikantha Phani and J. Woodhouse (Dept. of Eng., Univ. of Cambridge, Trumpington St., Cambridge CB2 1PZ, UK)

Identification of damping matrices from measured frequency response functions is a difficult and error-prone procedure. In practice, the assumption of proportional damping is frequently made without physical justification. Several researchers have suggested identification algorithms for the full damping matrix. A critical review is presented of algorithms which work in the frequency domain. Some novel approaches are included: one new algorithm is based on matrix perturbation theory, while others are hybrid combinations of earlier methods. The sensitivity of the algorithms to measurement noise and other commonly encountered difficulties in realistic measurements is addressed. Representative results from a simulation study comparing all the methods will be presented, leading to recommendations for the choice of identification algorithms in practice.

3:45

2pSA6. Boundary damping of flexural vibrations in beams and plates. Joel Garrelick (Applied Physical Sciences, Inc., 2 State St., Ste. 300, New London, CT 06320)

The damping of resonantly enhanced flexurally vibrating beams and plates is typically accomplished with surface treatments, viz. constrained or unconstrained damping layers. Alternatively, damping may be achieved at boundaries. In some sense this is a more fundamental approach in that for homogeneous plates, it is the boundaries that are solely responsible for resonant behavior. The theoretical performance of boundary treatments is unbounded, provided the treatment is full, that is continuous along the extent of the boundary. This is not the case however with partial coverage, where a portion of the boundary is left bare. The bounds for such treatments are explored in this paper. The treatment itself is defined in terms of a single bounce reflection coefficient (RC). It is found that for boundary damping in one dimension, viz., beams, the effective loss factor for individual modes is frequently invariant with either one or both ends treated, assuming RC constant. This is in contrast to the two-dimensional case, viz., thin plates, where analogous loss factor values are frequency dependent. Illustrative examples are presented and analyzed. [Work partially performed at CAA/Anteon Corp. and supported by NSWCCD and NSSC, Code 93R.]

4:00

2pSA7. What is the best method for structural damping identification? Part 2: Comparison of performance. Srikantha Phani and J. Woodhouse (Dept. of Eng., Univ. of Cambridge, Trumpington St., Cambridge CB2 1PZ, UK)

Damping identification in structural vibration from measured frequency response functions is a difficult task. Several methods have been proposed in the literature to identify the damping matrix. Some employ modal parameters such as natural frequencies, damping factors, and mode shapes, while others work directly from the frequency response function matrix. In practice, in the presence of noise and incompleteness of data, it is hard to guess how well any of these methods will perform. This paper reports a systematic simulation study to test the relative benefits/disadvantages of a wide range of approaches, existing and novel. Representative model systems have been chosen to test performance under various conditions of measurement noise, system complexity, damping type, modal overlap factor, and modal truncation. A very wide range of results is summarized by computing various norms of performance for all the methods studied. Recommendations are made for the best methods to extract reliable damping information from practical measurements.

4:15

2pSA8. Active noise control using damped resonant filters. Jesse B. Bisnette, Jeffrey S. Vipperman, and Daniel D. Budny (Dept. of Mech. Eng., Univ. of Pittsburgh, 648 Benedum Hall, Pittsburgh, PA 15261)

Active Noise Control (ANC) has been found to work well at low frequencies, thereby complementing passive noise control techniques. There are two distinct topologies for ANC: feedforward and feedback. Feedforward implementations work through the superposition of a primary and secondary sound field while feedback systems work by augmenting the dynamics of an enclosed sound field to add damping. Model-based feedback control systems are cumbersome to implement due to typical high modal density and simple output feedback systems are complicated for two reasons. First, perfectly collocated sensor (microphone) and actuator (loudspeaker) pairs are difficult to achieve, and second, the dynamics of the loudspeaker are destabilizing. Lane and Clark (1998) have reported compensating loudspeaker dynamics over a bandwidth to produce an approximate volume velocity source and demonstrated rate feedback acoustic control. Here, an acoustic control method using second order filters, which is analogous to positive position feedback (PPF) control in structural systems, is presented. The phase response of the loudspeaker is accounted for by incorporating negative feedback with either band-pass or high-pass filters, and the technique appears to have the same tolerance for actuator dynamics as for the structural case. Alternatively, the approximate volume velocity source could also be used.

4:30

2pSA9. An active flow control scheme for the reduction of cavity pressure in flow-excited Helmholtz resonators. Jong Beom Park and Luc Mongeau (Ray W. Herrick Labs., School of Mech. Eng., Purdue Univ., West Lafayette, IN 47907-2031)

Flows over cavities acting as flow-excited Helmholtz resonators feature the formation of strong vortices over the cavity orifice, which strength regulates the amplitude of the cavity pressure oscillations. Much is already known about this phenomenon, and models for predicting the magnitude and the frequency of the pressure oscillations are available. Various control devices and schemes have been developed to suppress the oscillations. In particular, active flow control using oscillated leading edge spoilers has been proposed as an alternative to passive spoilers, or flow injection systems. Little is known, however, about the detailed effects of such an active flow control system on the flow field. In the present study, a simple control loop was implemented to adjust the phase between the leading edge spoiler motion and the cavity pressure. The sound pressure inside a resonator was measured over a range of phase differences and spoiler motion amplitudes. It was found using flow visualization that optimally synchronized spoiler motion caused the vortices over the cavity orifice to be less concentrated. The suppression mechanisms were explained using a model based on vortex sound theory.

4:45

2pSA10. When things go crunch: Gap length effects on a magnetorheological tunable vibration absorber. Anne-Marie Albanese and Kenneth Cunefare (Georgia Inst. of Technol., Atlanta, GA 30332)

A tunable vibration absorber (TVA), where the tunable element is a magneto-rheological (MR) elastomer spring, has demonstrated up to a 460% change on tuning frequency. A frequency increase is associated with a significant decrease in the springs static equilibrium length, referred hereto as a crunch. The crunch is caused by magnetic attractive forces across the spring. The spring, an iron-doped silicone gel, is placed between two halves of a low-carbon steel loop. One loop half behaves as an

absorber mass, and has approximately 200 turns of magnet wire around it. Driving current through this wire generates a magnetic flux around the steel path and through the MR spring. Beyond a threshold, the magnetic field induces a large frequency shift in the absorber, with a crunch observed across the elastomer. The crunch can occur when the static equilibrium length is unconstrained by geometry, thus a magnetic attractive force shortens the spring length. The relationship between the magnitude of the crunch and the frequency shift will be presented. Additionally, the impact of different initial MR spring lengths on the frequency behavior will be considered. Finally, the frequency variability achievable by MR-spring-based TVAs with and without the crunch will be assessed.

TUESDAY AFTERNOON, 29 APRIL 2003 ROOM 205, 1:00 TO 5:20 P.M.

Session 2pSC

Speech Communication: Man and Machine in Cocktail Parties

Pierre L. Divenyi, Cochair
Speech and Hearing Research, VA Medical Center, 150 Muir Road, Martinez, California 94553

Daniel P. W. Ellis, Cochair
Department of Electrical Engineering, Columbia University, Mail C4712, 500 West 120th Street, Room 1312, New York, New York 10027

Chair's Introduction—1:00

Invited Papers

1:05

2pSC1. Some components of the "cocktail-party effect," as revealed when it fails. Pierre L. Divenyi and Brian Gygi (Speech and Hearing Res., Veterans Affairs Northern California Health Care Systems and East Bay Inst. for Res. and Education, Martinez, CA 94553)

The precise way listeners cope with cocktail-party situations, i.e., understand speech in the midst of other, simultaneously ongoing conversations, has by-and-large remained a puzzle, despite research committed to studying the problem over the past half century. In contrast, it is widely acknowledged that the cocktail-party effect (CPE) deteriorates in aging. Our investigations during the last decade have assessed the deterioration of the CPE in elderly listeners and attempted to uncover specific auditory tasks, on which the performance of the same listeners will also exhibit a deficit. Correlated performance on CPE and such auditory tasks arguably signify that the tasks in question are necessary for perceptual segregation of the target speech and the background babble. We will present results on three tasks correlated with CPE performance. All three tasks require temporal processing-based perceptual segregation of specific non-speech stimuli (amplitude- and/or frequency-modulated sinusoidal complexes): discrimination of formant transition patterns, segregation of streams with different syllabic rhythms, and selective attention to AM or FM features in the designated stream. [Work supported by a grant from the National Institute on Aging and by the V.A. Medical Research.]

1:35

2pSC2. Monaural speech segregation. DeLiang Wang (Dept. of Computer & Information Sci. and Ctr. for Cognit. Sci., The Ohio State Univ., Columbus, OH 43210, dwang@cis.ohio-state.edu) and Guoning Hu (The Ohio State Univ. Biophys. Prog., Columbus, OH 43210)

Speech segregation from a monaural recording is a primary task of auditory scene analysis, and has proven to be very challenging. We present a multistage model for the task. The model starts with simulated auditory periphery. A subsequent stage computes midlevel auditory representations, including correlograms and cross-channel correlations. The core of the system performs segmentation and grouping in a two-dimensional time-frequency representation that encodes proximity in frequency and time, periodicity, and amplitude modulation (AM). Motivated by psychoacoustic observations, our system employs different mechanisms for handling resolved and unresolved harmonics. For resolved harmonics, our system generates segments—basic components of an auditory scene—based on temporal continuity and cross-channel correlation, and groups them according to periodicity. For unresolved harmonics, the system generates segments based on AM in addition to temporal continuity and groups them according to AM repetition rates. We derive AM repetition rates using sinusoidal modeling and gradient descent. Underlying the segregation process is a pitch contour that is first estimated from speech segregated according to global pitch and then adjusted according to psychoacoustic constraints. The model has been systematically evaluated, and it yields substantially better performance than previous systems.

2:05

2pSC3. How does spatial hearing affect cocktail party conversations? Barbara G. Shinn-Cunningham (Boston Univ., 677 Beacon St., Boston, MA 02215, shinn@cns.bu.edu)

Although spatial cues are not a dominant factor in auditory scene analysis by humans, spatial separation of a target talker and an interfering sound source often increases the intelligibility of the target. A large portion of this improvement arises from a simple acoustic effect: separating the target and interferer generally increases the target-to-interferer energy ratio at one ear. When the target is near threshold, binaural processing provides an additional small, but important improvement. Finally, in conditions where both target and interferer are audible but are difficult to segregate, spatial separation can improve the ability to stream the sources and interpret the target message. The echoes and reverberation present in everyday environments influence all of these factors, altering the target-to-interferer energy ratio, the effectiveness of binaural processing, and the ability to stream the sources. A number of studies will be reviewed to demonstrate how these different factors influence speech intelligibility and affect cocktail party conversations in everyday environments. [Work supported by the Air Force Office of Scientific Research and the Alfred P. Sloan Foundation.]

2:35

2pSC4. Active audition for humanoid robots that can listen to three simultaneous talkers. Hiroshi G. Okuno (Dept. of Intelligence Sci. and Technol., Grad. School of Informatics, Kyoto Univ., Sakyo, Kyoto 606-8501, Japan, okuno@i.kyoto-u.ac.jp) and Kazuhiro Nakadai (Kitano Symbiotic Systems Project, JST, M-31 6-31-15 Jingumae, Shibuya, Tokyo 150-0001, Japan)

The direction-pass filter (DPF) separates sounds originating from a particular direction by using a pair of microphones embedded in each ear of humanoid robot. DPF first extracts harmonic structures from each channel, finds a corresponding pair on right and left channels, and then calculates their interaural phase difference (IPD) and interaural intensity difference (IID). These IPD and IID are matched with reference data obtained by HRTF or by the geometrical relation to determine the sound source direction. The direction obtained by face detection may be used as a candidate for the direction. Finally, all subbands from the direction are collected to synthesize a wave form by inverse FFT. The allowance of collection depends on the direction; narrow (10 deg) at center, while wide (30 deg) at the periphery. This property is called ''auditory fovea'' and is exploited by DPF actively to improve performance of sound source separation. In addition, a humanoid actively turns its head toward the speaker to listen better. Real-time DPF is implemented by distributed processing with five PCs. Preliminary experiments of active audition in speech recognition of three simultaneous utterances of digits in a normal room is also reported. [Work supported by JSPS.]

3:05–3:20 Break

3:20

2pSC5. Machine recognition of sounds in mixtures. Daniel Ellis (Columbia Univ., 500 W. 120th St., Rm. 1312, New York, NY 10027) and Jon Barker (Sheffield Univ., Sheffield S1 4DP, UK)

A machine simulation of human auditory perception must be able to recognize and classify individual sound sources. The most successful technique for sound classification is the statistical pattern recognition approach employed in speech recognizers; however, in most practical cases, this approach assumes that the entire (monaural) signal represents the source to be classified. Realistic ''cocktail-party'' scenes, composed of multiple, overlapping sources with comparable energies, do not come close to meeting this assumption. A more workable assumption is to treat each time-frequency cell as representing a single source, and to use missing-data techniques to perform recognition using only a subset of the cells. This precludes the use of cepstral features (which depend on every frequency component), but is otherwise practical. The problem then becomes finding the ''present data mask'' that indicates which cells are to be considered during classification of a particular source. We will present a system based on these principles, with applications both to speech recognition in dynamic, noisy backgrounds, and also to nonspeech sounds such as alarms that can occur at very poor signal-to-noise ratios.

3:50

2pSC6. How is harmonicity used in grouping speech sounds? Chris Darwin (Exp. Psych., Univ. of Sussex, Brighton BN1 9QG, UK, cjd@biols.susx.ac.uk)

This paper asks how a common property of voiced speech sounds harmonicity is used by the auditory system to improve the perception of speech in the presence of simultaneous competing sounds. We present data from three different experimental paradigms concerned, respectively, with the combination of sounds across different ears, different frequency regions, and different times. The first set of experiments qualify the conclusion that sounds from the same harmonic series fuse into a single object when presented to different ears. The second impose limits on the ability of harmonicity to combine information across different frequency regions. The third demonstrate the utility of continuity of pitch (compared with a vocal-tract size) in maintaining attention to a single sound source. Elucidating the mechanisms by which we segregate speech from background sounds requires proper consideration both of the structure of the speech signal and of the auditory system through which it passes.

4:20

2pSC7. Scene analysis without spectral analysis? Alain de Cheveigne (Ircam–CNRS, 1 place Igor Stravinsky, Paris 75004, France, cheveign@ircam.fr)

Auditory scene analysis is often described in terms of grouping stimulus components. Components, once grouped, are assigned to one source or another [A. S. Bregman, *Auditory Scene Analysis* (MIT, Cambridge, MA, 2002)]. The actual grouping must operate on whatever representation is available within the auditory nervous system. An obvious hypothesis is that correlates of individual stimulus components are created by peripheral spectral analysis. However, peripheral frequency resolution is limited. The number of resolved partials is between 5 and 8 for a harmonic complex in isolation, but resolution must necessarily be less good for the interleaved components of concurrent sources. Source amplitudes are rarely equal, and partials of a weaker source must be particularly hard to resolve. The question is thus: given the paucity of resolved elements to group, how does the auditory system perform the grouping? A number of possibilities will be reviewed. One is that partials not resolved peripherally are somehow resolved centrally (a modern version of the ''second filter'' hypothesis). Another is that scene analysis does not operate by grouping resolved elements, but instead by modifying directly unresolved entities, for example by time-domain processing.

4:50

2pSC8. Issues in the use of acoustic cues for auditory scene analysis. Albert S. Bregman (Psych. Dept., McGill Univ., 1205 Doctor Penfield Ave., Montreal, QC H3A 1B1, Canada)

Issues concerning auditory scene analysis (ASA) raised by the previous speakers will be discussed: (1) Disorders of ASA in humans can tell us about the weighting of cues in ASA. (2) The apparent weakness of spatial cues for ASA may simply show that they interact strongly with other ASA cues (c.f., recent research in the author's lab). (3) The power of harmonic relations among partials as a grouping cue is not guaranteed, but depends on many other factors. (4) Abstract models of ASA may require the peripheral auditory system to carry out analyses that are questionable, based on current psychophysical and physiological findings. Is this where psychologists and computational ASA (CASA) modelers part company? (5) The ''old-plus-new heuristic,'' one of the most potent ASA mechanisms, is neglected by existing CASA models. (6) The different roles of bottom-up and top-down processes (e.g., in ''exclusive allocation'' of sensory evidence) should be reflected in models. (7) Should the output of a CASA system be the reconstructed signal of a single source, as a front end to a recognition system, or should grouping mechanisms merely form an interacting part of a larger system that outputs a higher-level description (e.g., a series of words)?

TUESDAY AFTERNOON, 29 APRIL 2003 ROOM 203, 1:00 TO 2:35 P.M.

Session 2pSP

Signal Processing in Acoustics and Physical Acoustics: Subspace Methods for Acoustical Imaging II

Sean K. Lehman, Cochair
Lawrence Livermore National Laboratory, L-154, 7000 East Avenue, Livermore, California 94550

David H. Chambers, Cochair
Lawrence Livermore National Laboratory, L-154, P.O. Box 808, Livermore, California 94551-5508

Chair's Introduction—1:00

Invited Papers

1:05

2pSP1. A model subspace method for matched field source localization in a shallow-water acoustic waveguide. Peter M. Daly (SAIC Ocean Systems Operation, 1710 SAIC Dr., M.S. 1-11-15, McLean, VA 22102, peter.m.daly@saic.com)

Model subspace methods for parameter estimation can yield promising results when temporal or spatial variability dominate the model inputs. Application of Gaussian signal detection theory, summarized by Van Trees, is applied to the localization problem. Model inputs, such as the sound velocity profile in water, bottom properties, and array position, rather than being deterministic, are characterized using a Gaussian random vector. This, in turn, yields a random signal model which is represented as a vector space. Detection and localization via a maximum-likelihood estimator becomes a question of whether the received signal fits the subspace defined by the random signal model.

1:35

2pSP2. Interference cancellation in a waveguide using signal subspace decomposition. Paul Hursky, Michael B. Porter (Sci. Applications Intl. Corp., 10260 Campus Point Dr., San Diego, CA 92121), and John P. Ianniello (Sci. Applications Intl. Corp., Mystic, CT 06355)

When the presence of a strong interferer hides a quieter target, an eigenanalysis of the data covariance matrix (also called a principal component analysis) may separate the quieter target from the strong interferer. However, the strong interferer may occupy more than a single principal component when the data covariance matrix is estimated over a time interval during which the strong interferer has moved between multiple spatial resolution cells of the receive array. Also, when the sources to be distinguished are of comparable strength, the principal components fail to isolate the sources, forming mixtures of the sources instead. Results will be presented of applying this technique to the SWellEx-96 experiment data, recorded on both vertical and horizontal arrays, in a shallow water environment off the coast of San Diego. Projections of the principal eigenvectors onto beams and matched-field processing range-depth cells will be shown to indicate to what degree a principal component analysis isolates the different sources. Strategies for coping with source motion will be discussed.

Contributed Papers

2:05

2pSP3. Application of Hilbert space decomposition to acoustical inverse problems. Sean K. Lehman (Lawrence Livermore Natl. Lab., L-154, 7000 East Ave., Livermore, CA 94550)

In a recently developed theory, the forward integral acoustic scattering operator is cast into the formalism of a Hilbert space operator which projects the continuous space scattering object into the discrete measurement space. By determining the singular value decomposition (SVD) of the forward scattering operator, one obtains optimal, orthonormal bases for each of these spaces in the form of the singular vectors. In formulating the inverse scattering problem, it is best to express the unknown object distribution in terms of an expansion of the singular vectors. Using this expansion, the best reconstruction, in a LMS sense, can be obtained from measured scattered field data. We present reconstruction results using this new theory. [Work performed under the auspices of the U.S. Department of Energy by University of California Lawrence Livermore National Laboratory under Contract No. W-7405-Eng-48.]

2:20

2pSP4. Acoustical imaging of spheres above a reflecting surface. David Chambers and James Berryman (Lawrence Livermore Natl. Lab., P.O. Box 808, L-154, Livermore, CA 94551)

An analytical study using the MUSIC method of subspace imaging is presented for the case of spheres above a reflecting boundary. The field scattered from the spheres and the reflecting boundary is calculated analytically, neglecting interactions between spheres. The singular value decomposition of the response matrix is calculated and the singular vectors divided into signal and noise subspaces. Images showing the estimated sphere locations are obtained by backpropagating the noise vectors using either the free space Green's function or the Green's function that incorporates reflections from the boundary. We show that the latter Green's function improves imaging performance after applying a normalization that compensates for the interference between direct and reflected fields. We also show that the best images are attained in some cases when the number of singular vectors in the signal subspace exceeds the number of spheres. This is consistent with previous analysis showing multiple eigenvalues of the time reversal operator for spherical scatterers [Chambers and Gautesen, J. Acoust. Soc. Am. **109** (2001)]. [Work performed under the auspices of the U.S. Department of Energy by the University of California, Lawrence Livermore National Laboratory under Contract No. W-7405-Eng-48.]